PLANET
Networking & Communication


# SIP IP Phone

## VIP-254T/VIP-254PT

# User's manual

### Version 1.0

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## CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

## WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

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# Revision

User’s Manual for PLANET SIP IP Phone:

Model: VIP-254T/VIP-254PT

Rev: 1.0 (2008, May)

Part No. EM-VIP254V1.0

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# Chapter 1 1

# Introduction

## Overview

Meeting the next-generation Internet telephony service demands, PLANET Technology provides feature-rich, toll-quality Internet telephony service solutions. With 802.3af Power over Ethernet (PoE) IP Phone－VIP-254PT. And the VIP-254T is the cost-effective SIP IP Phone; the VIP-254 series are SIP 2.0 (RFC3261) compliant with SIP digest authentication supports.

The VIP-254T / VIP-254PT ("IP Phone" in the following term) features high-quality speakerphone technology, and includes an easy-to-use speaker on/off button and call hold/transfer buttons for various voice services.
The IP Phone has additional features such as built-in PPPoE/DHCP clients, password-protected machine management, LCD menu display, 3-way conference keys, hands-free speakerphone, incoming message indicator, and user-intuitive web administration system.
The IP Phone is self-contained, service-integrated, intelligent phone features offering, and powerful voice processing power. The IP Phone can effortlessly deliver toll voice quality equivalent to the regular SIP protocol connections utilizing cutting-edge Quality of Service, echo cancellation, comfort noise generation (CNG) and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the IP Phone allow users to install in an existing network location without interfering with desktop PC network connections.
Besides, the IP Phones are ideal solution for office / home use as well as installation for Internet Telephony Service Provider (ITSP) from leading vendors. It's the delivery platform for IP voice services that makes benefit from the VoIP technologies in your daily life.

There are models for VIP-254T/VIP-254PT and there are:
**VIP-254T:** SIP IP Phone
**VIP-254PT:** 802.3af PoE SIP IP Phone

## Product Features

- **Simple Installation and administration**

  Configuration of the **IP Phone** can be performed in minutes via the LCD menu keypad, or web interfaces. Using the built-in LCD display, the **IP Phone** offers user-friendly configuration guidelines, machine operation status, call status displays, and incoming call identification.

- **Feature-rich keypad IP Phone**

  The **IP Phone** integrates a high-quality speakerphone with the Call Hold, Forward, Transfer and Waiting functions and also provides advanced telephone features, such as 3-way conference key, incoming call history indicator in a much more convenient and functional manner than traditional telephone sets.

- **Dynamic IP address assignment, and voice communication**

  The **IP Phone** can act as a PPPoE/DHCP client, automatically obtaining an IP address for Internet access.

- **Various field applications compliant**

  The **IP Phone** is capable of handling peer-to-peer and SIP proxy / IP PBX registration, authentication to interact with major IP PBX/SIP gateway/IP Phone in the market. The IP Phone offers the most flexibility and interoperability with PLANET and 3rd party VoIP vendors, allowing the deployment of both simple and complex VoIP networks such as ITSP, PC-to-Phone/Phone-to-PC or enterprise VoIP environments.

- **Standards compliant**

  The VIP-254T / VIP-254PT complies with SIP 2.0 (RFC3261), interoperates with 3rd party SIP voice gateways/terminal/software as well as other PLANET VoIP products. Supported Voice codecs and VoIP technologies are: G.723, G.729ab, G.711u-law/a-law; Voice Activity Detection (VAD), and the Confort Noise Generation (CNG).

## VoIP Features

- SIP 2.0 (RFC3261) compliant
- Peer-to-Peer / SIP proxy calls
- Voice codec support: G.711, G.723.1, G.726, G.729A, G.729B
- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation
- In band and out-of-band DTMF support

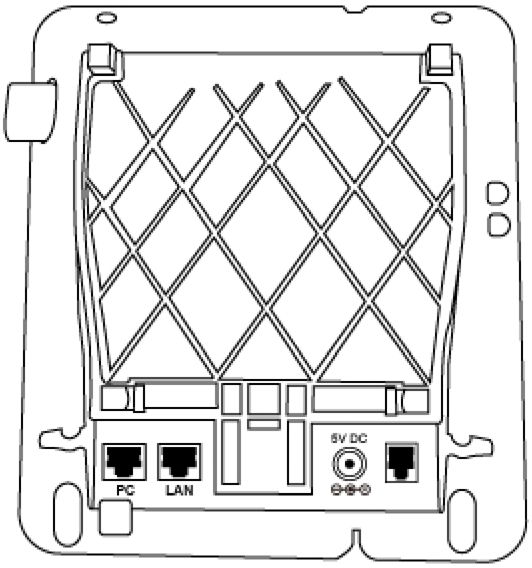
## Package Content

The contents of your product should contain the following items:
VoIP IP Phone
Power adapter
Quick Installation Guide
User's Manual CD
RJ-45 cable x 1

# Physical Details

The following figure illustrates the front/rear panel of IP Phone.

## Rear View



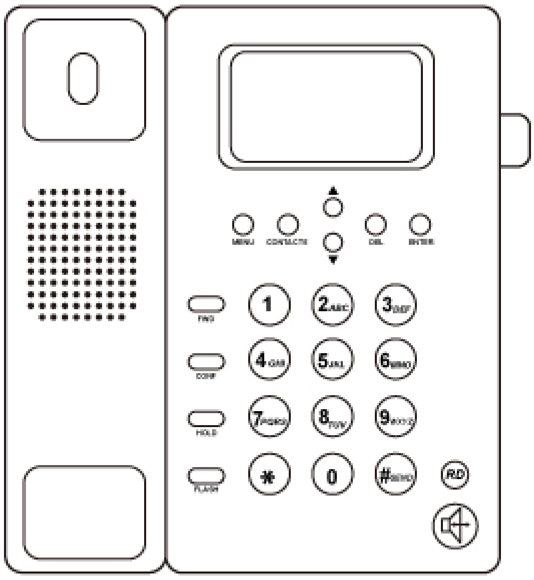

**Rear Panel of VIP-254T/VIP-254PT**

| | | |
|---|---|---|
| **1** | **PC** | RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable |
| **2** | **LAN** | RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.<br><br>The **LAN** interface also can be connected with 802.3af PoE switch or converter for power supply (**VIP-254PT**) |
| **3** | **DC 5V** | 5V DC Power input outlet |
| **4.** | **Handset** | RJ-11 connector, connected directly to the Handset. |

**ⓘNote**

For VIP-254PT, either PoE or AC adapter can be deployed at one time

## Front View and Keypad function



**Front Panel of VIP-254T/VIP-254PT**

### Keypad Description

| | | |
|---|---|---|
| **1** | **LCD Display** | Menu and all status shall be displayed for users. |
| **2** | **MENU** | To bring out the menu selection while IP Phone is in idle state. |
| **3** | **▲ ▼** | This is Up ▲ / Down ▼ key and volume setting when off-hook off.<br>Show the calls history when on-hook. |
| **4** | **ENTER** | To be used as confirm configuration or enter sub-menu. |
| **5** | **CONTACTS** | Enter the phone book selection. |
| **6** | **FLASH** | To transfer an active call (incoming call answered or outgoing call accepted) to another devices. |
| **7** | **CONF** | Press this button can make conference function. |
| **8** | **FWD** | To carry out forward function. |
| **9** | **DEL** | Press to delete digits when at configuration mode or input phone numbers.<br>Press to mute sounds when at talk mode. |

| | | |
|---|---|---|
| **10** | **RD** | Press to dial the last dialed number when the IP Phone is off-hooked. |
| **11** | **Handfree** | To switch between the usage of the handset and the speaker devices. |
| **12** | **Hold** | To hold the conversation. |

# Chapter 2

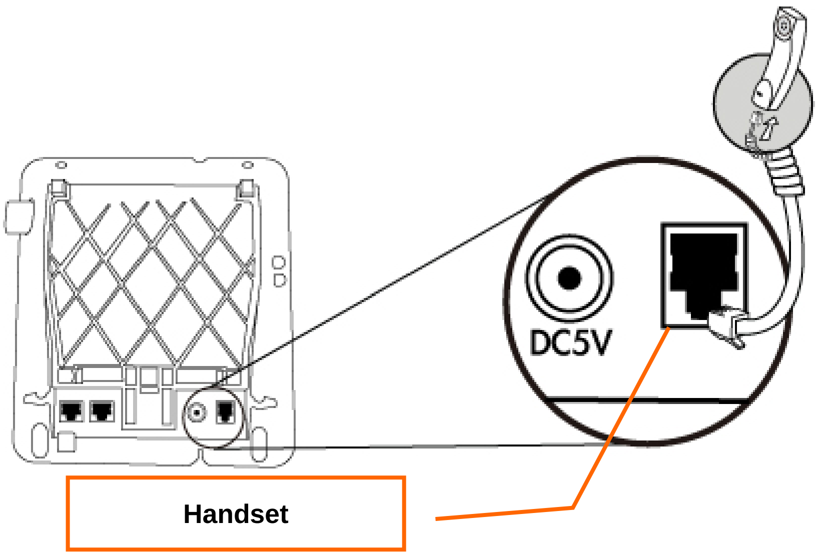
# Preparations & Installation

## Physical Installation Requirement

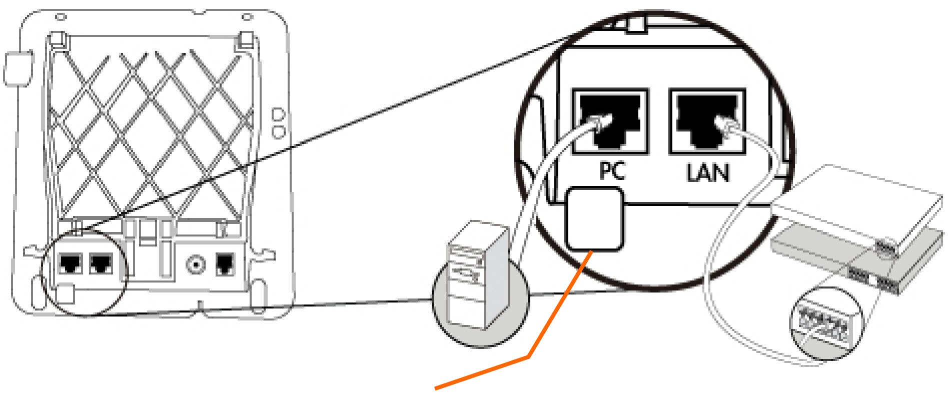
**VIP-254T:** SIP IP Phone (2 x RJ-45)

**VIP-254PT:** 802.3af PoE SIP IP Phone (2 x RJ-45, 1 x PoE for LAN interface)

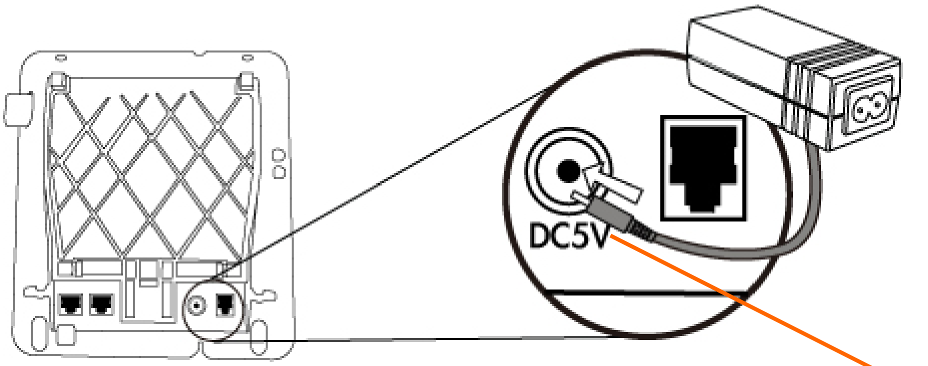
### Step 1: Connecting Handset




### Step 2: Connecting Power AC Power and Network



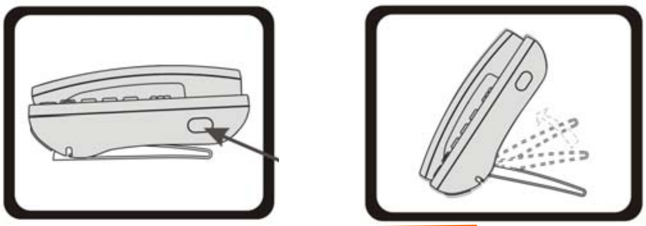

**Plug the Ethernet cable into the back of the base station. Plug the other end of the Ethernet cable into your already prepared network connection.**




**Power Adapter (5V DC)**

**NOTE: Use only the power adapter shipped with the unit to ensure correct functionality.**

## Step 3: Adjust the stand angle.



**Press and hole the button of right side to change the stand mount angle.**

| | | |
|---|---|---|
| **1** | **PC** | RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable |
| **2** | **LAN** | RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.<br><br>The **LAN** interface also can be connected with 802.3af PoE switch or converter for power supply (**VIP-254PT only**) |
| **3** | **5V DC** | 5V DC Power input outlet |
| **4** | **Handset** | RJ-11 connector, connected directly to the Handset. |

**Note**

1. For VIP-254PT, either PoE or AC adapter can be deployed at one time

## Administration Interface

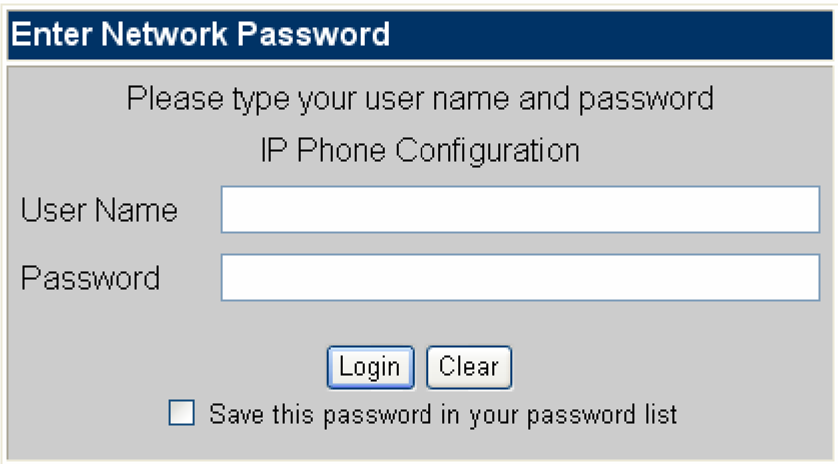
The IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration. Key pad administration also available for simple configuration.

### Web configuration access:

To start IP Phone web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0.0 or higher with Java support

Default IP address of IP Phone is **192.168.0.1**. You may now open your web browser, and insert ***http://192.168.0.1*** in the address bar of your web browser to logon IP Phone web configuration page. IP Phone will prompt for logon username/password, please enter: ***root*** / **null** (no password) to continue machine administration.




**Note**

In order to connect machine for administration, please locate your PC in the same network segment (192.168.0.x) of IP Phone. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

# Chapter 3

# Network Service Configurations

## Configuring and monitoring your IP Phone from web browser

The IP Phone integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

### Overview on the web interface of IP Phone

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

To start IP Phone web configuration, you must have one of these web browsers installed on computer for management

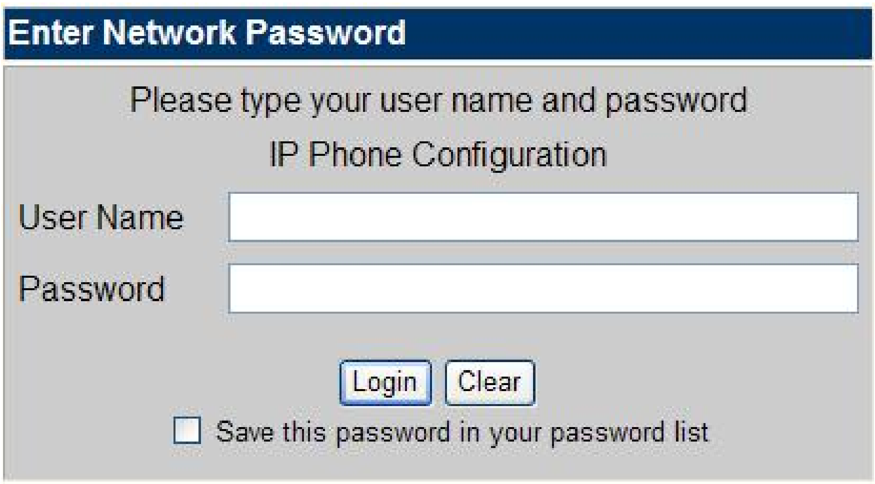
- Microsoft Internet Explorer 6.0.0 or higher with Java support

### Manipulation of IP Phone via web browser

**Log on IP Phone via web browser**

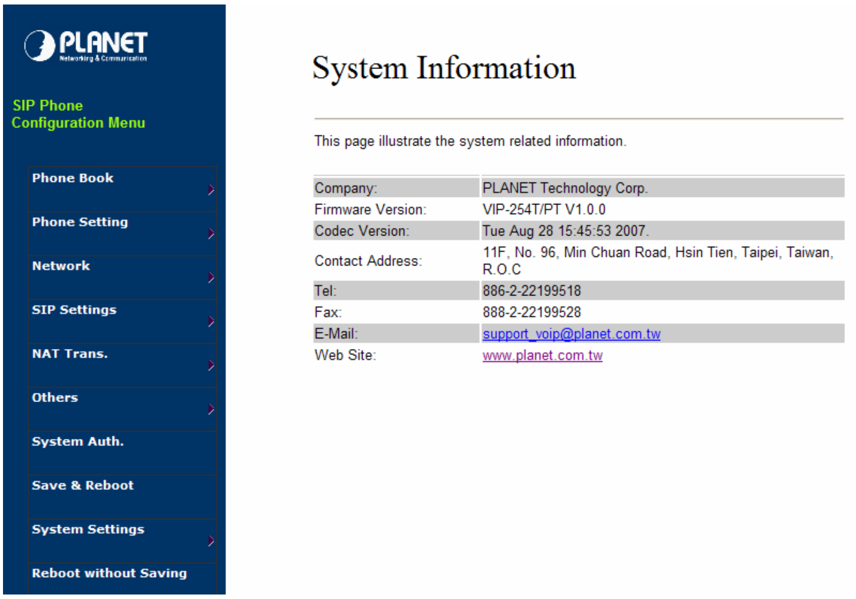
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon IP Phone web configuration page.

IP Phone will prompt for logon username/password: ***root*** / ***null (without password)***




***IP Phone log in page***

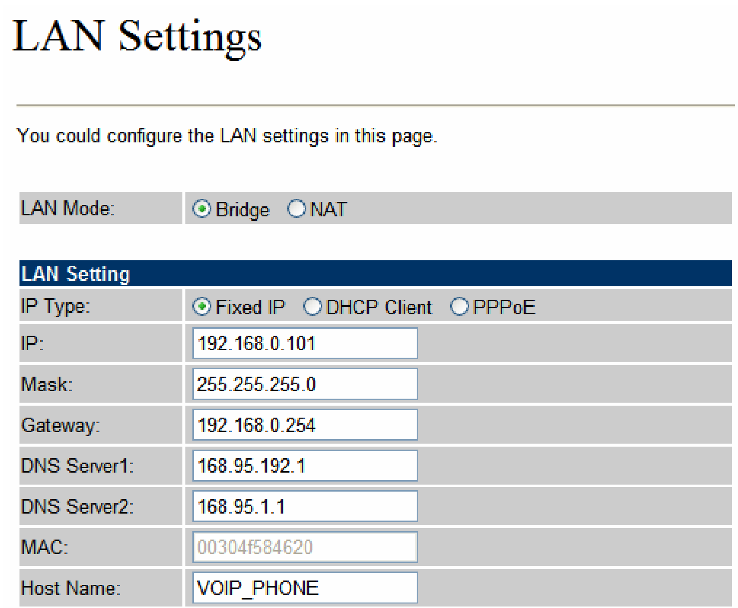
When users login the web page, users can see the IP Phone system information like firmware version, company…etc in this main page.




***IP Phone main page***

## LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP in the address bar. After logging on machine with username/password (default**: root / no password**), browse to "**Network**" --> "**LAN Settings**" configuration menu:




### Parameter Description

| | |
|---|---|
| **IP address** | LAN IP address of IP Phone<br><br>**Default**: 192.168.0.1 |

**Subnet Mask** LAN mask of IP Phone

**Default**: 255.255.255.0

**Default Gateway** Gateway of IP Phone

**Default**: 192.168.0.254

After confirming the modification you've done, Please click on the **Submit** button to apply settings and browse to "**Save & Reboot**" menu to reboot the machine to make the settings effective.

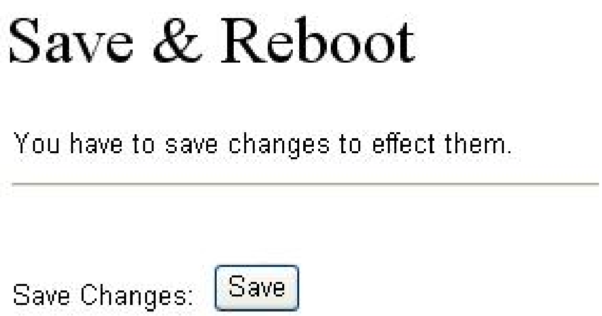
| Connection Type | Data required. |
|---|---|
| **Fixed IP** | In most circumstances, it is no need to configure the DHCP settings. |
| **DHCP client** | The ISP will assign IP Address, and related information. |
| **PPPoE** | The ISP will assign PPPoE username / password for Internet access, |

ⓘ Hint

Please consult your ISP personnel to obtain proper PPPoE/IP address related information, and input carefully.
If Internet connection cannot be established, please check the physical connection or contact the ISP service staff for support information.

### Save Modification to Flash Memory

Most of the IP Phone parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the IP PHone, to save the parameters into Flash ROM and let it take effective forever, please remember to press the **Save & Reboot** button after you modify the parameters.

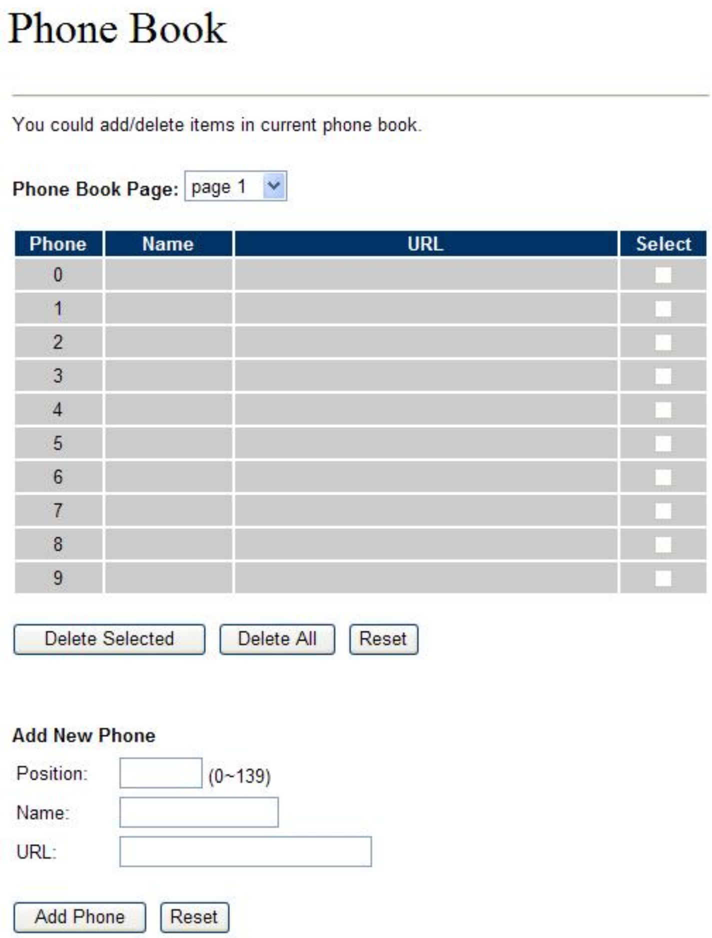


# Chapter 4

# VoIP IP Phone Configurations

## Phone Book settings

IP Phone can set up 140 records of Phone Book. User can make calls via **Phone Book** feature of IP Phone.




| Field | Description |
|---|---|
| **Phone Book Page** | The default is Page 1. It can select Page1 ~ Page 14 to look round Phone Book records. |
| **Phone** | The record number from 0 ~ 139, it can set up 140 records in total. |
| **Name** | The name of Phone Book records, it only can input numerals. |
| **URL** | Fill in the outgoing number (Line Number) or IP address. |
| **Select** | To select this record. |

If you need to add a phone number into the Phone Book list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

If you want to delete all phone numbers, you can click “Delete All” button.

## For Example:

### Phone Book

You could add/delete items in current phone book.

**Phone Book Page:** page 1

| Phone | Name | URL | Select |
|---|---|---|---|
| 0 | 301 | 301@192.168.1.2 | ☐ |
| 1 | 206 | 17476433364 | ☐ |
| 2 | 202 | 192.168.1.2:5062 | ☐ |
| 3 | | | |
| 4 | | | |
| 5 | | | |
| 6 | | | |
| 7 | | | |
| 8 | | | |
| 9 | | | |

Delete Selected | Delete All | Reset

**STEP 1:**

IP Phone had added the above phone numbers. User press **Phone Book** button from keypad then the LCD screen will show below:

```
Search:          [   3]
```

**STEP 2:**

Press OK button to enter the Phone Book menu. The LCD screen will show the Phone Book records pervious made.

```
00   202
01   206
```

**STEP 3:**

Selecting the recorder you want to dial and press OK button. It sill show the detail information as below:

| | |
|---|
| 202 |
| 192.168.1.2:5062 |

**STEP 4:**

Pick up the telephone handset or press Handfree button to dial to this telephone.

| | |
|---|
| IP Dialing..          1 |
| 192.168.1.2:5062 |

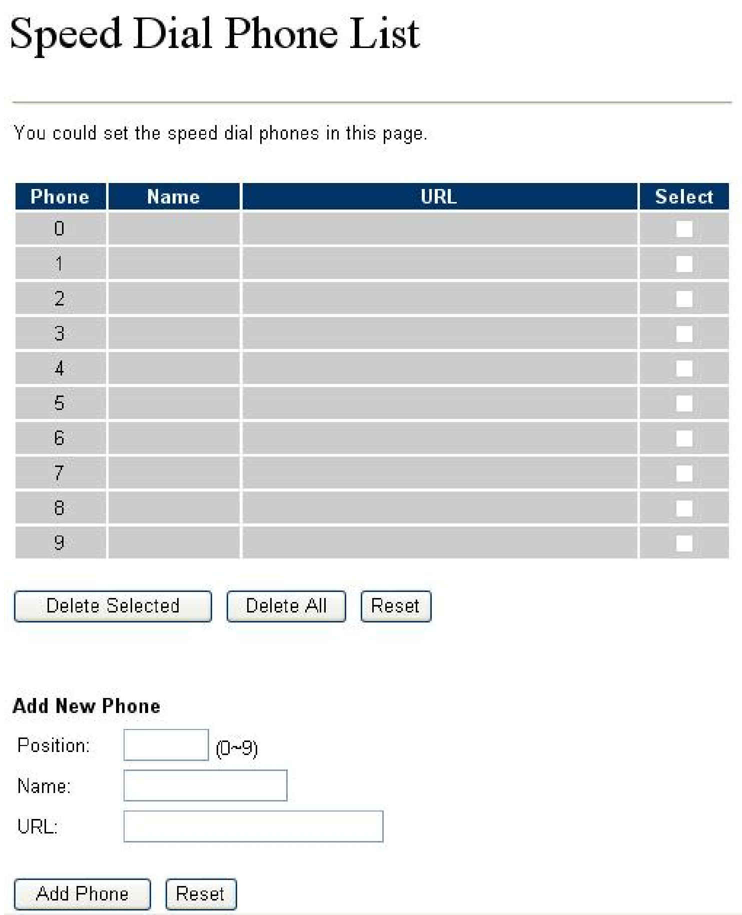
## Speed Dial settings

In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) and follow the “#” key.

If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

If you want to delete all phone numbers, you can click “Delete All” button.



## Call Forward

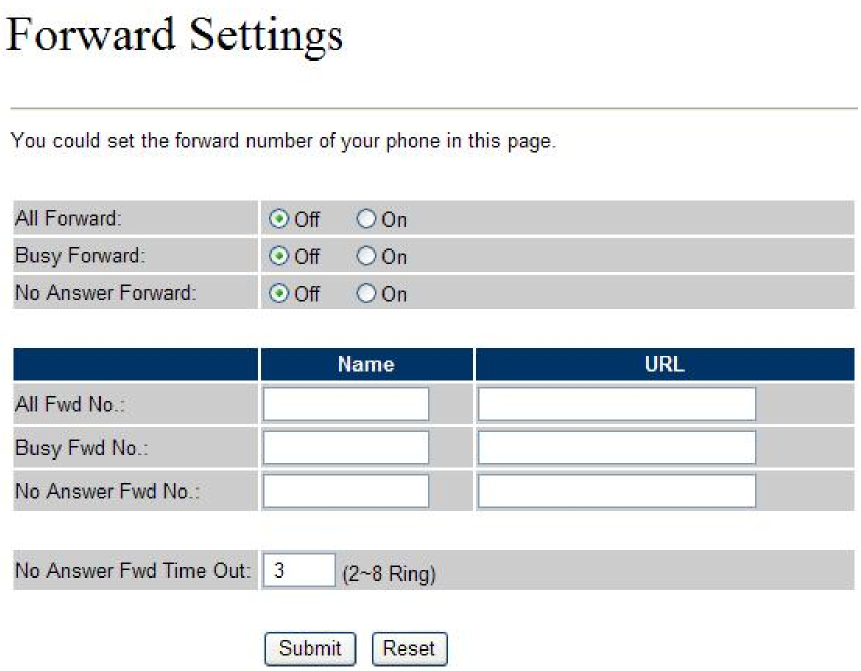
This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

**All Forward**: All incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

**Busy Forward**: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

**No Answer Forward**: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

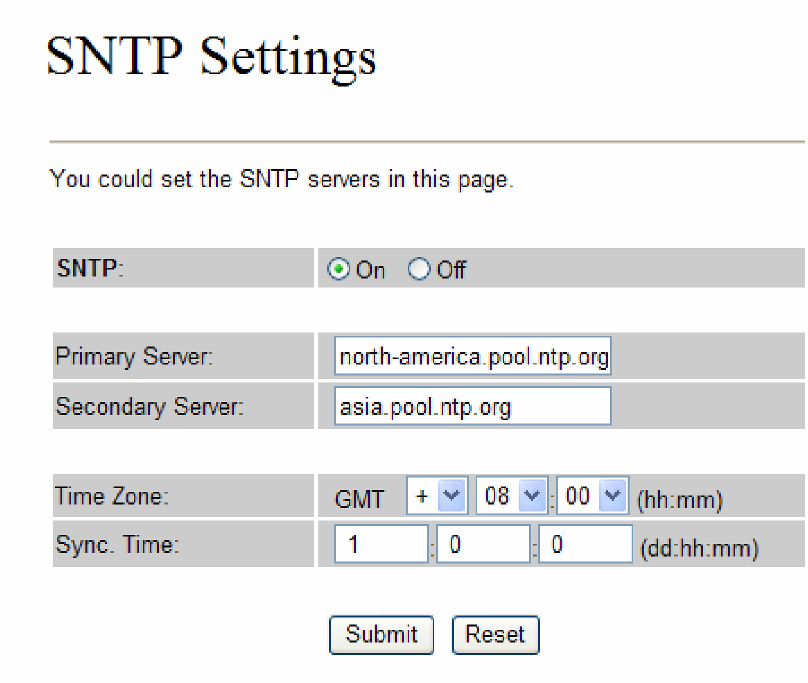
When you finished the setting, please click the Submit button.




***Call Forward function for VIP-254T/VIP-254PT***

## SNTP settings

This page defines the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.




## Volume Setting

This page defines the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

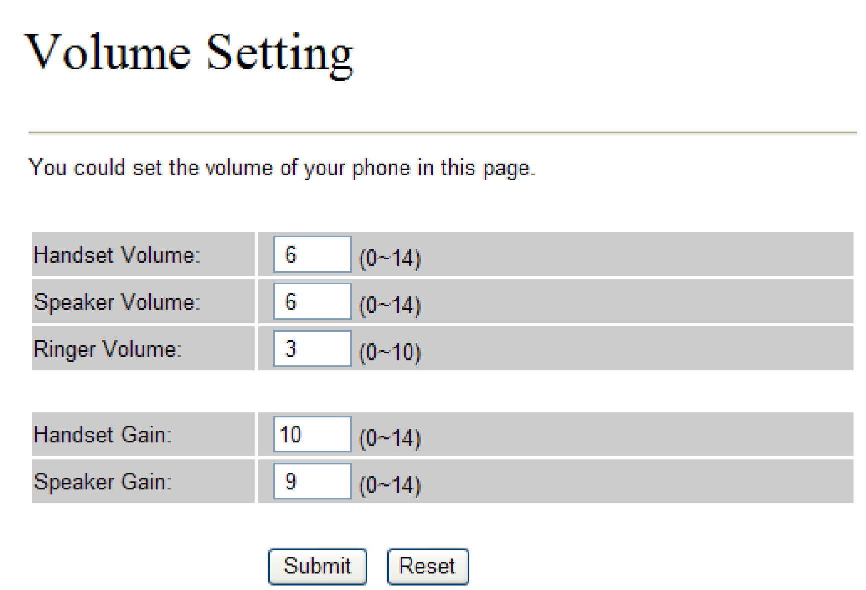
**Handset Volume** is to set the volume for you can hear from the handset.(Handfree mode)

**Speaker Volume** is to set the volume for you can hear from the speaker.

**Ringer Volume** is to set the ringer volume for you can hear.

**Handset Gain** is to set the volume send out to the other side's handset.

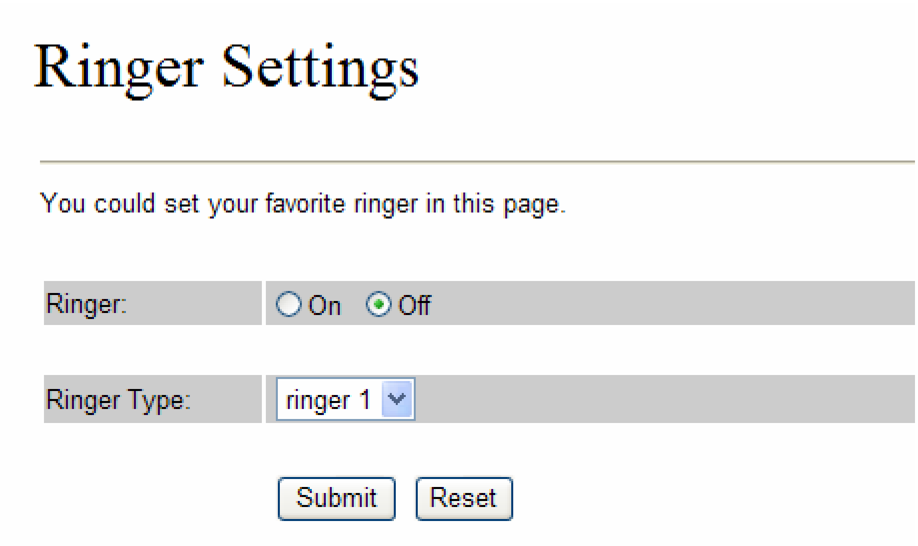
**Speaker Gain** is to set the volume send out to the other side's handset from the microphone. (Handfree mode)




***Volume Settings for VIP-254T/VIP-254PT***

## Ringer Setting

This page defines the user can set the tinkle of bells when someone ring your IP Phone. If want to set ringer, it need to enable Ringer function and select the Ringer Type you wanted. There are four Ringer Types can be chosen. When you finished the setting, please click the Submit button.



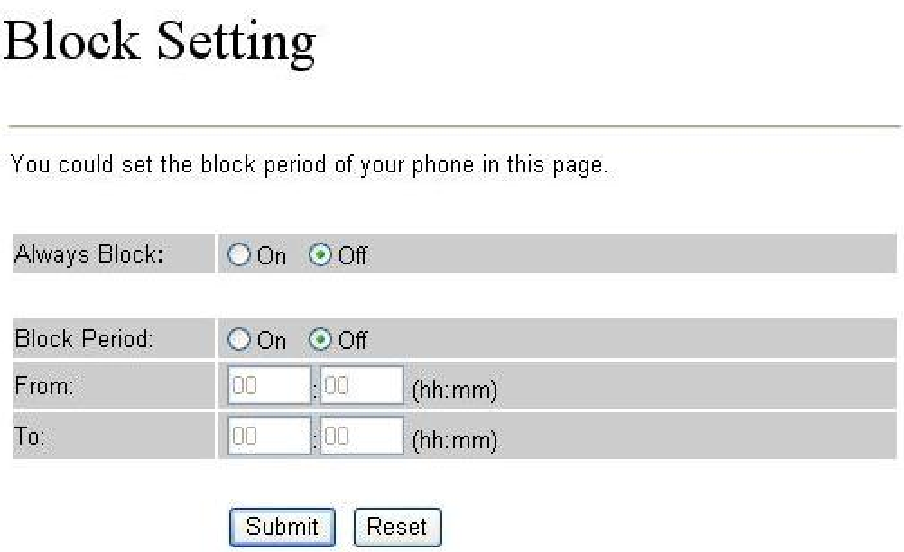

## Block Setting

This page defines the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

**Always Block**: All incoming call will be blocked until disable this feature.

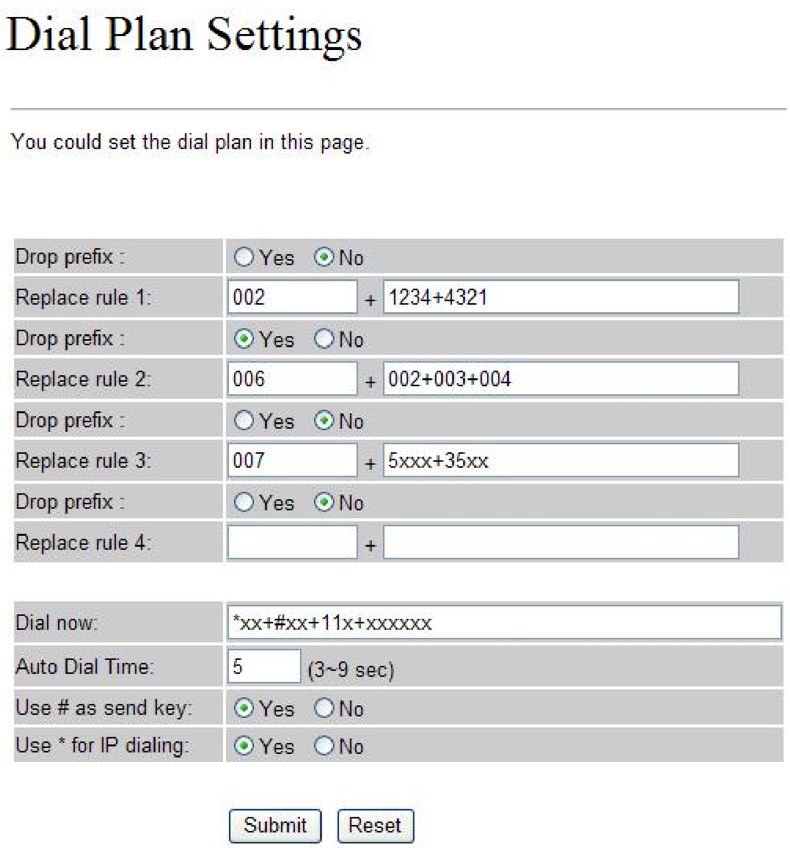
**Block Period**: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

When you finished the setting, please click the Submit button.



## Dial Plan Settings

This page defines the Dial Plan Setting function. This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.




| Field | Description |
|---|---|
| **Drop Prefix** | The rule of add or replace code. If setup as No, it will add the prefix number prior to the identification number. If setup as Yes, it will replace the identification number. |
| **Replace rule** | The prefix number. It only accept the numeral and the max length is 8. |
| **+** | The identification number. It can accept the numeral or symbol and the max length is 40.<br>■ Symbol: It only accept the [+], [x]<br>■ +: It means as "or". For example, [123+456+334+5xx] even if [123 or 456 or 334 or 5xx]<br>■ x: It is equal to 0~9. For example, [5xx] even if the number begin 5. |
| **Dial Now** | If the dialing number are match with this field, it will dial out and need not to press the "#" key to end the dialing. It accepts the numeral or symbol, and the max length are 124.<br>ⓘ**Note**: The starting number can't be the "0". For example, if the number is "0xxxx", because the starting number is "0", so that the |

| | |
|---|---|
| | system will ignore this dial plan. |
| **Auto Dial Time** | Stop dialing after seconds then send dial number out. |
| **Use # as send key** | If setup as Yes, the system sill stop to receive the dialing number when receive the [#] key. The system also will to determine the Auto Dial Time, it will carry out the calling if there isn't receive the digit after the Auto Dial Time.<br>If setup as No, the system just according to the Auto Dial Time to determine the end time. |
| **Use * for IP dialing** | If setup as Yes, the system will look on [*] as [.]. For example, if dial the "192*168*0*100#", it will dial out as "192.168.0.100#".<br>If setup as No, it just look on [*] as [*]. For example, if dial the "700*#", it will dial out as "700*#". |

### Descriptions of example:

**Example_1:** Drop prefix: **No**, Replace rule 1: **002**, +: **1234+4321** (No limit the digit length)

1. If the dialing number is start as "**1234**", it will add the 002 at begin. The real dialing number is [**0021234…**].
2. If the dialing number is start as "**4321**", it will add the 002 at begin. The real dialing number is [**0024321…**].

**Example_2:** Drop prefix: **Yes**, Replace rule 2: **006**, +: **002+003+004** (No limit the digit length)

1. If the dialing number is start as "**002**", it will replace 002 by 006. The real dialing number is [**006…**].
2. If the dialing number is start as "**003**", it will replace 003 by 006. The real dialing number is [**003…**].

**Example_3:** Drop prefix: No, Replace rule 3: **007**, +: **5xxx+35xx** (Has limit the digit length)

1. If the dialing number start as "**5**" and follow 3 digits, it will add the 007 at begin. The real dialing number is [**0075xxx**].
2. If the dialing number start as "**35**" and follow 2 digits, it will add the 007 at begin. The real dialing number is [**00735xx**].

**Example_4:** Dial Now: ***xx+#xx+11x+xxxxxx**

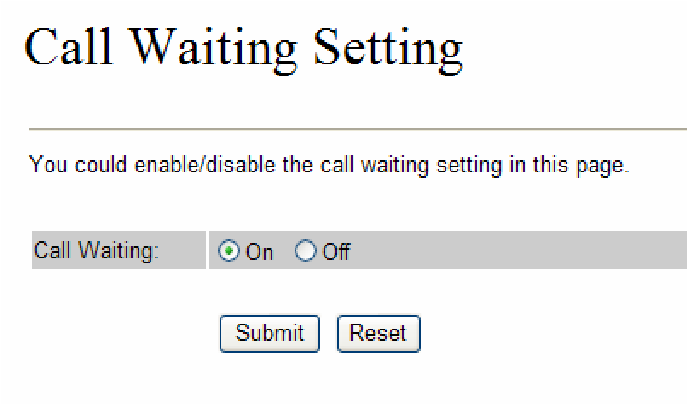
1. If the dialing number is match with the rule of "***xx**", it will send out the dialing number directly. For example, ***00/ *01/ *02…*99**.
2. If the dialing number is match with the rule of "**#xx**", it will send out the dialing number directly. For example, **#00/ #01/ #02…#99**.
3. If the dialing number is match with the rule of "**11x**", it will send out the dialing number directly. For example, **111/ 112/ 113…119**.
4. If the dialing number is match with the rule of 8 digits, it will send out the dialing number directly. For example, **12345678**.

ⓘ**Note**

If enable the **Routing** function and the dialing number is match with **Routing rule**, machine will carry out the Routing function and to skip over the below **Drop prefix** and **Replace rule** functions.

## Call waiting Settings

When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear any notice.

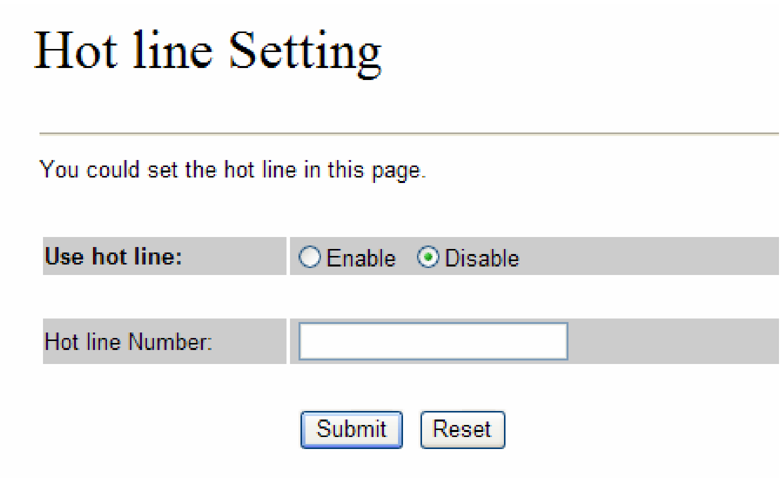



## Hot line Settings

This page defines the Hot line setting in this page. When user pick up the handset, the device will call to the specific number automatically.

**Use Hot Line:** Click Enable to carry the Hot line function out.

**Hot line number:** The hot line number, it can input the IP address or registration number.



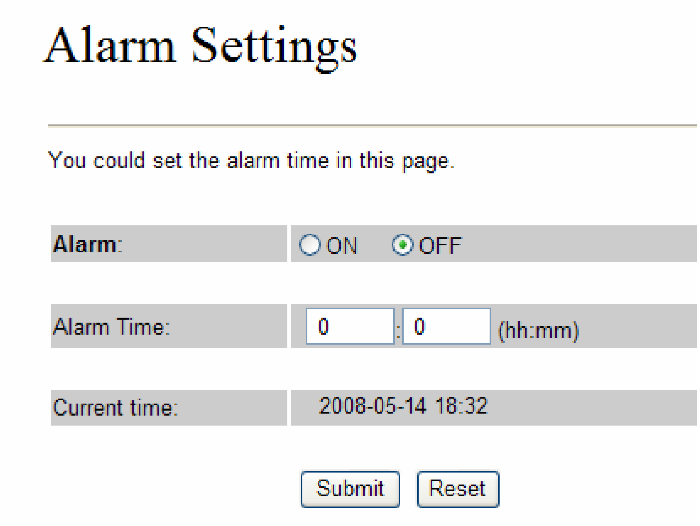

## Alarm Settings

This page defines the Alarm setting in this page. It provides the alarm function, and it can set up the Alarm Time to get the telephone ringed up every day.

**Alarm:** The default is Off. If set up as On, the telephone will ringed up at the specific time.

**Alarm Time:** It can set up the system prompt time with 24 hours.

**Current time**: The next alarm time.




## LAN Settings

This page defines the LAN setting in this page.

**LAN Mode:** The default is Bridge mode, and it also provides NAT mode.

- **Bridge:** When set as is mode, the LAN and PC ports are in the same network segment.
- **NAT:** The LAN and PC ports are in the different network segment, and PC port could enable the DHCP Server function to allot the IP address.

**IP Type:** The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes.

- **Fixed IP:** It could setup the IP address manual.
- **DHCP Client:** It will acquire the IP address automatically.
- **PPPoE:** It will use the PPPoE connection method.

**IP:** The IP address

**Mask:** The sub net address

**Gateway:** The default gateway address

**DNS Type:** The default is Fixed mode, it could setup the DNS mode to manual or auto detection.

**DNS Server1:** The default is 168.95.192.1, it could setup the first DNS server address.

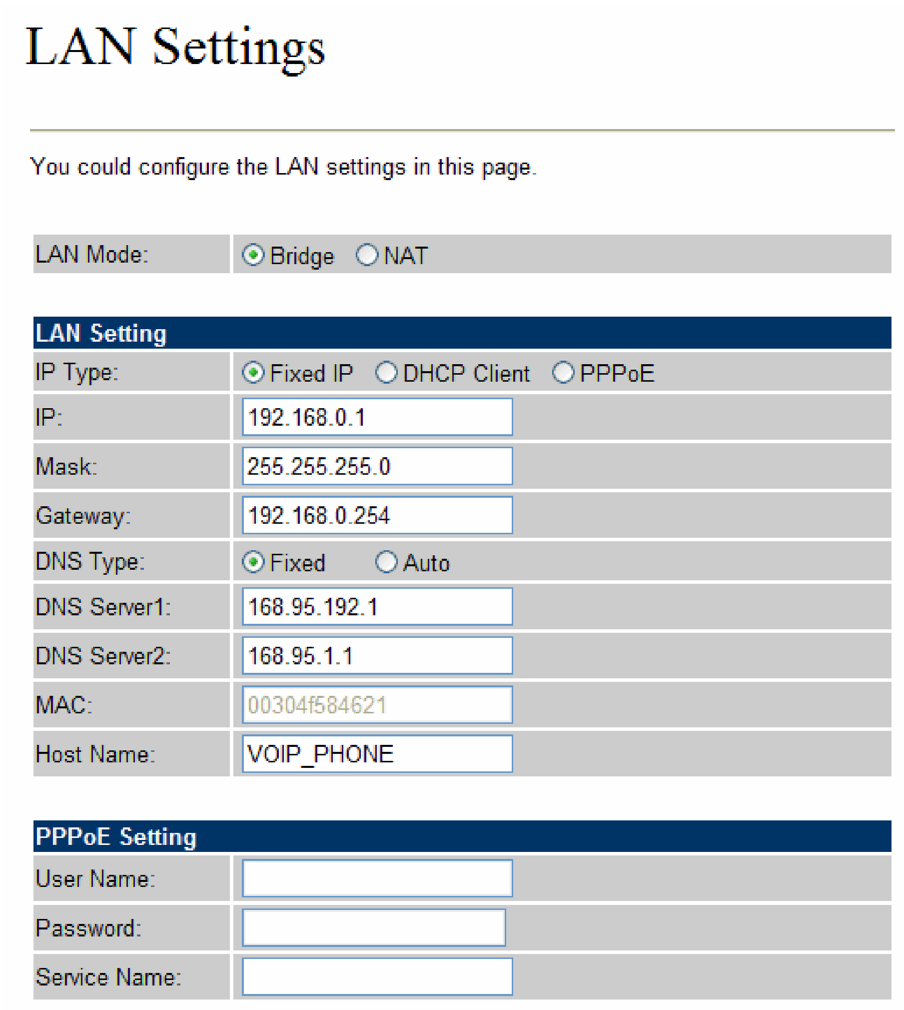
**DNS Server2:** The default is 168.95.1.1, it could setup the second DNS server address.

**MAC:** The MAC of LAN port

**Host Name:** The product model

**User Name:** The PPPoE connection account name. It could inpout numeral or character, the maximum date length are 63.

**Password:** The PPPoE connection account password. It could inpout numeral or character, the maximum date length are 63.




## PC Settings

This page defines the PC setting in this page.

**IP:** The IP address of PC port. (In the **Bridge** mode, the **Default IP**: 192.168.123.1)

**Mask:** The sub net address. (**Default**: 255.255.255.0)

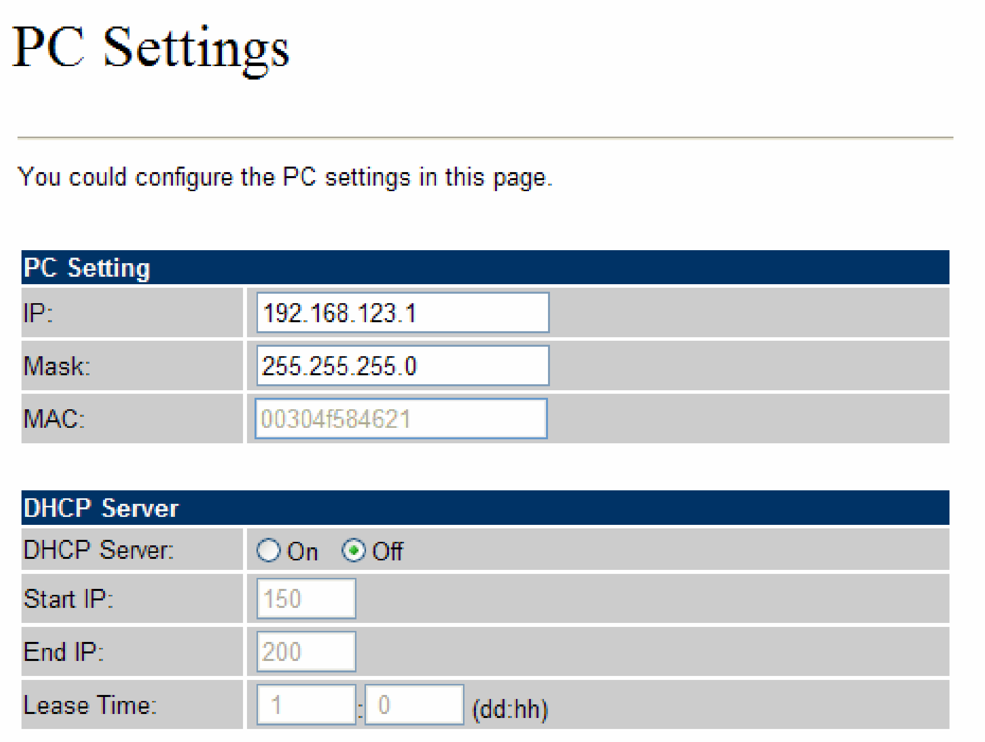
**MAC:** The MAC of PC port

**DHCP Server:** It will allot the IP address automatically when enabke this function.

**Start IP:** Start IP of lease table

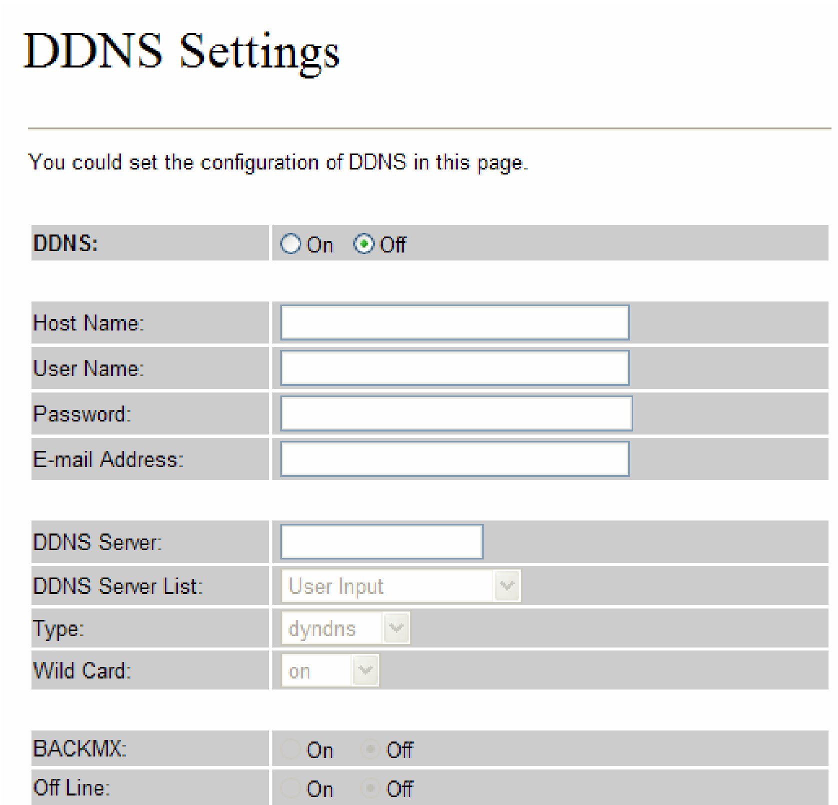
**End IP:** End IP of lease table. Network device connecting to the PC port can dynamic obtain the IP in the range between start IP and end IP

**Lease Time:** DHCP server lease time




## DDNS Settings

This page defines the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.



## VLAN Settings

This page defines the VLAN setting in this page. This function needs to co-operate with network devices which have VLAN function.

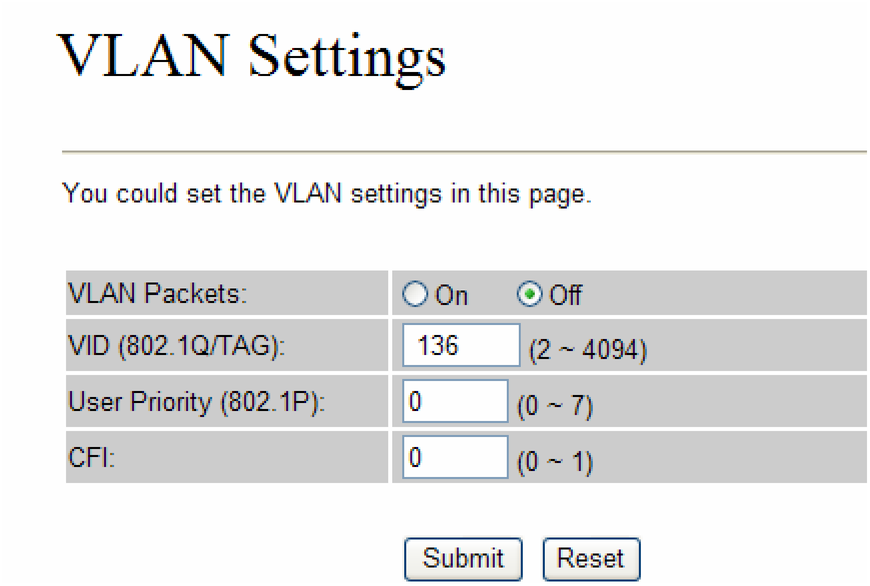
**VLAN Packets:** If setup as On, it could receive VLAN messages.

**VID (802.1Q/TAG):** Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 2~4094.

**User Priority (802.1P):** To setup the user priority.

**CFI:** To indicate the Canonical Format.

- If CFI=1, it means the header label include RIF field, and the NCIF flag valus of RIF will to decide the MAC address is Canonical Format or Non-Canonical Format in frame information.
- If CFI=0, it means the header label does not include RIF field, and the MAC address is Canonical Format in frame information.

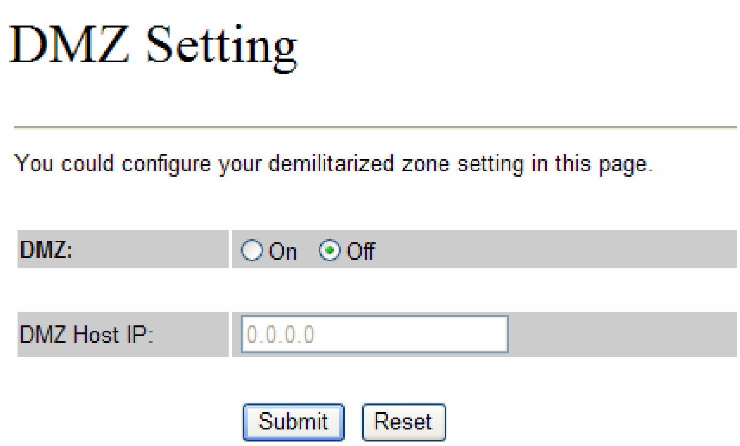



## DMZ Settings

This page defines the DMZ setting in this page.

**DMZ:** If setup as On, all of packets (expect SIP packets) will send to the specific IP address.

**DMZ Host IP:** The DMZ host IP address.



## Virtual Server

This page defines the Virtual Server setting in this page. You could define 24 virtual service information in this page. When you finished the setting, please click the Submit button.

**Virtual Server Page:** There are total page1 to page 3. It could choose the page which want to go over.

**Num:** The serial number. There are total 24 records from Num 0 to 23.

**Enable:** The activate status. The default is Disable, this record will been activate if enable.

**Protocol:** The TCP or UDP communication protocol.

**Internal Port:** For corresponding the internal port.

**External Port:** For corresponding the external port.

**Server IP:** To input the Server IP address.

# Virtual Server Settings

You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].

**Virtual Server Page:** page 1

| Num | Enable | Protocol | In Port | Ex Port | Server IP | Select |
|---|---|---|---|---|---|---|
| 0 | ☐ | | | | | ☐ |
| 1 | ☐ | | | | | ☐ |
| 2 | ☐ | | | | | ☐ |
| 3 | ☐ | | | | | ☐ |
| 4 | ☐ | | | | | ☐ |
| 5 | ☐ | | | | | ☐ |
| 6 | ☐ | | | | | ☐ |
| 7 | ☐ | | | | | ☐ |

Enable Selected | Delete Selected | Delete All | Reset

**Add Virtual Server**

Server IP:

Protocol: TCP

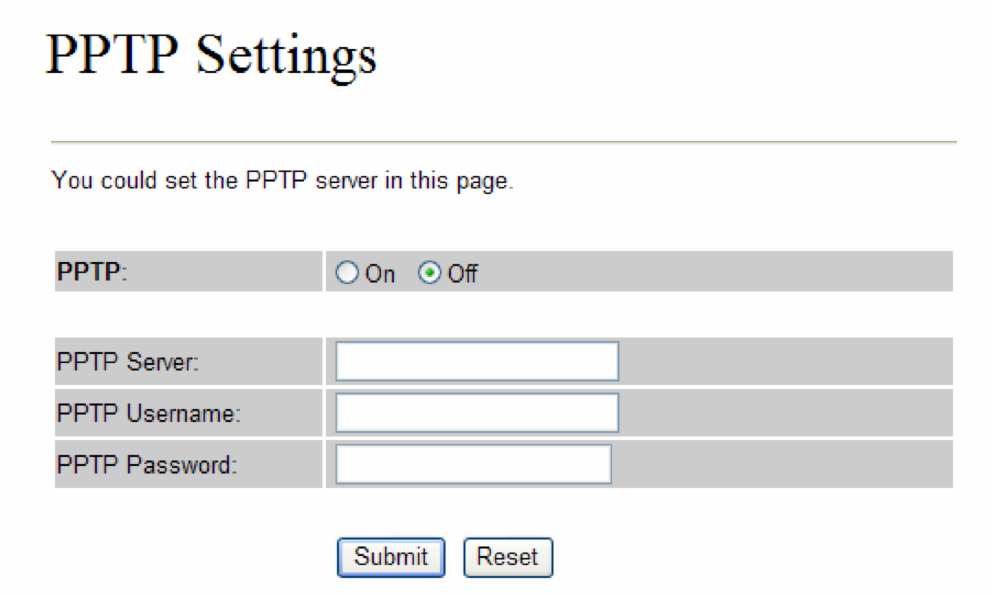
Internal Port Start: Internal Port End:

External Port Start: External Port End:

Add Server | Reset

## PPTP Settings

This page defines the PPTP setting in this page. You could setup the PPTP Server connection information. When you finished the setting, please click the Submit button.




## Service Domain Settings

This router comes with the built-in firewall based on the advanced technology of Stateful Packet

In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the Phon. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

**Display Name**: you can input the name you want to display.

**User Name**: you need to input the User Name get from your ISP.

**Register Name**: you need to input the Register Name get from your ISP.

**Register Password**: you need to input the Register Password get from your ISP.

**Domain Server**: you need to input the Domain Server get from your ISP.

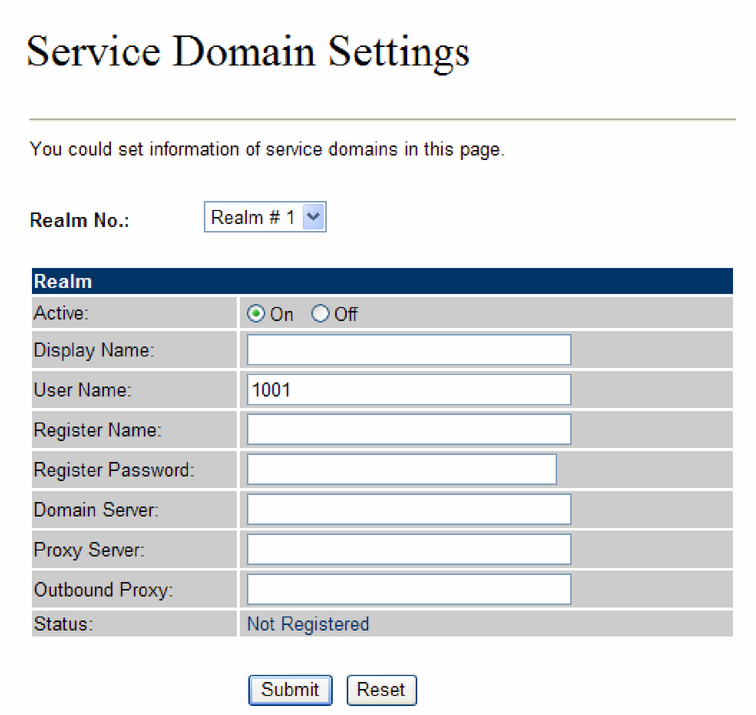
**Proxy Server**: you need to input the Proxy Server get from your ISP.

**Outbound Proxy**: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

You can see the Register Status in the Status item. If the item shows “Registered”, then your Phone Adapter is registered to the ISP, you can make a phone call direcly.

If you have more than one SIP account, you can following the steps to register to the other ISP.

When you finished the setting, please click the Submit button.




**ⓘNote**:

IP Phone can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input the switch code.

**Realm switch code:**

**1***: Realm 1

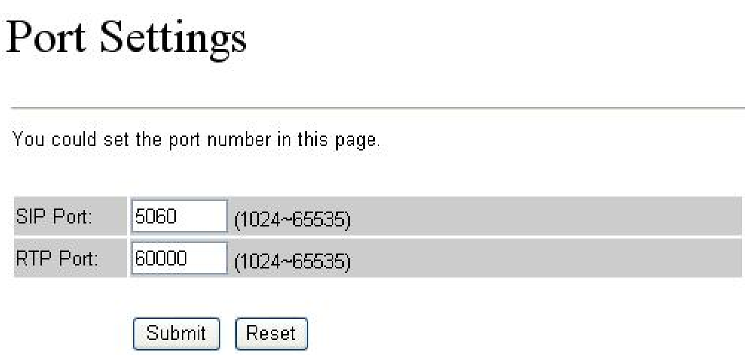
**2***: Realm 2

**3***: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

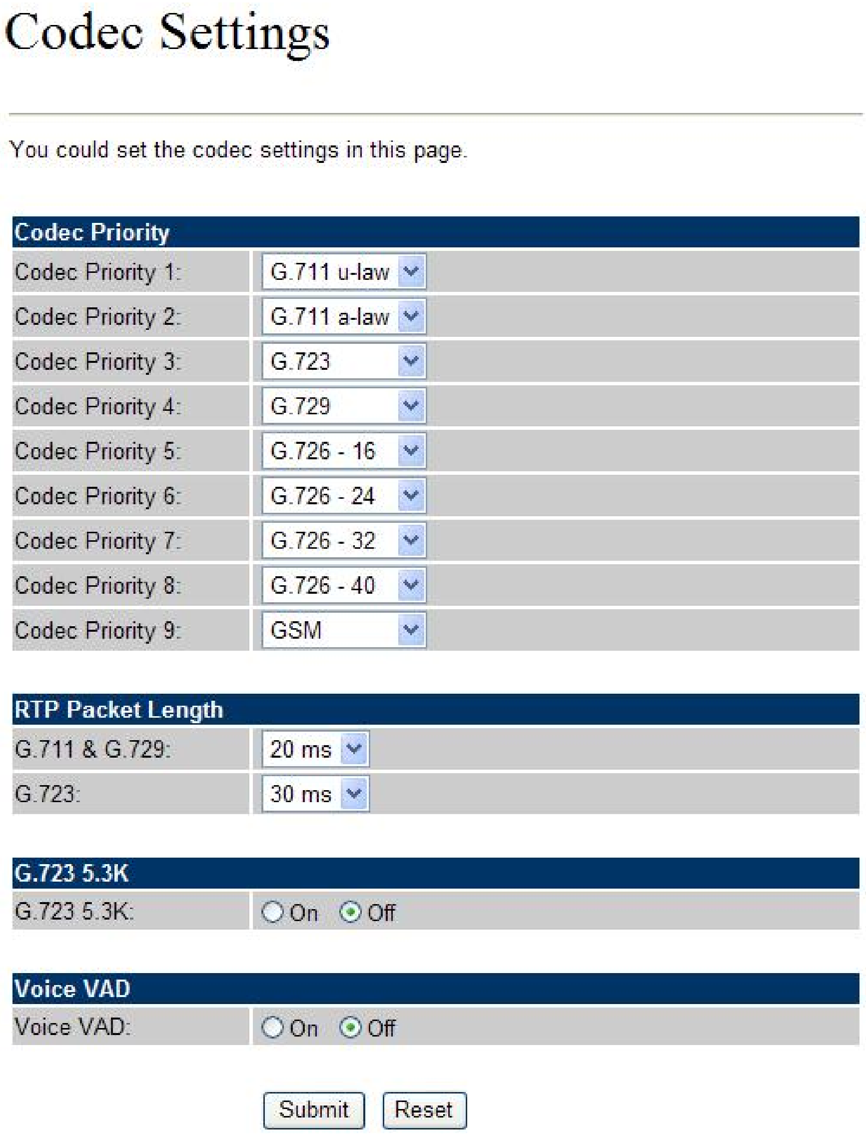
## Port Settings

This page defines the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.



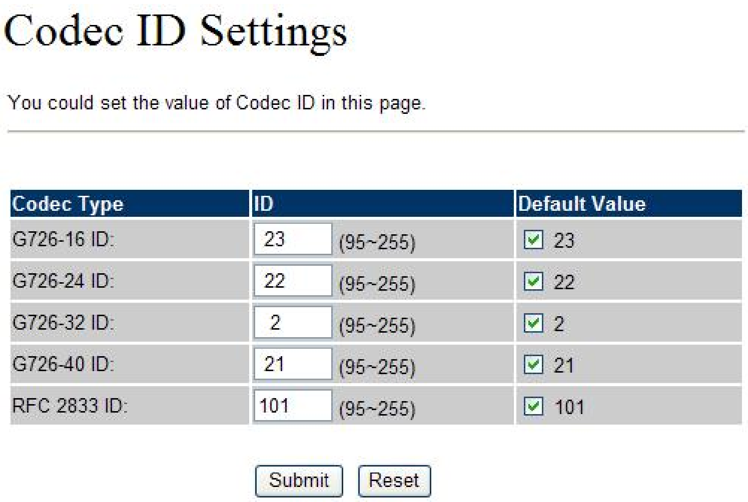
## Codec Settings

This page defines the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.



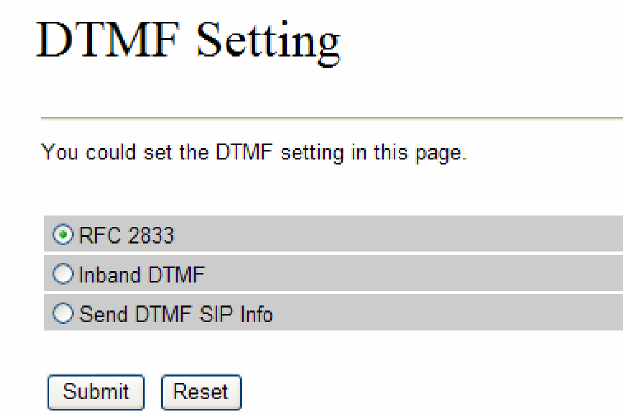

## Codec ID Setting

This page defines the Codec ID. Sometimes 2 VoIP device with different Codec ID will cause the interoperability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use then you can change your Codec ID. When you finished the setting, please click the Submit button.



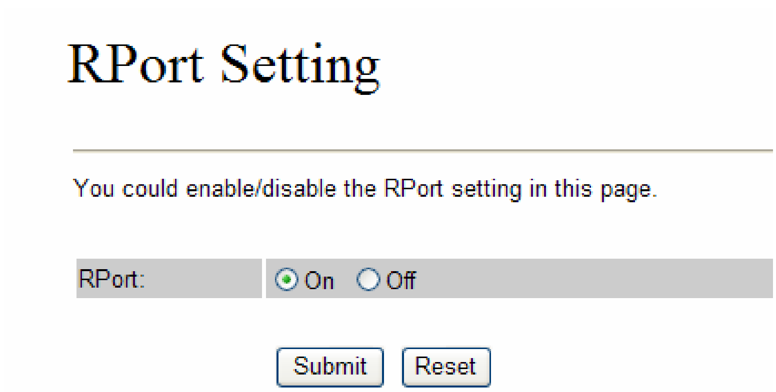

## DTMF Settings

This page defines the DTMF parameters. Yyou can setup the InBand DTMF, 2833 Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.




## RPort Settings

This page defines the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



## Other Settings

This page defines the Hold by RFC, Voice/SIP QoS and other settings in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

**Hold by RFC:** The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.

**Voice QoS (Diff-Serv):** The Voice QoS feature.

**SIP QoS (Diff-Serv)**: The SIP QoS feature.

The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

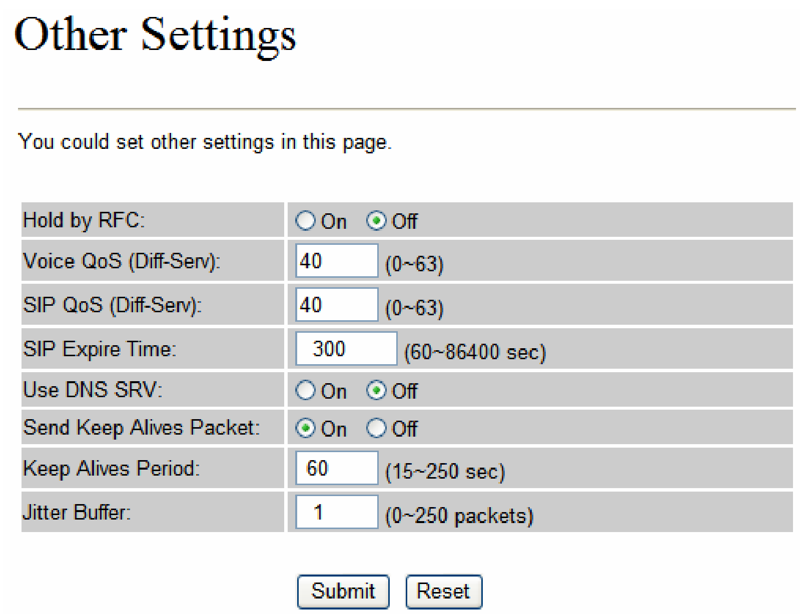
**SIP Expire Time:** To setup the registration interval time.

**Use DNS SRV:** The default is disable, and use DNS SRV mode. Set enable to use DNS to SRV mode to search the host information.

**Send Keep Alives Pcaket:** To deliver the packets on a regular time schedule to keep NAT port could open continued.

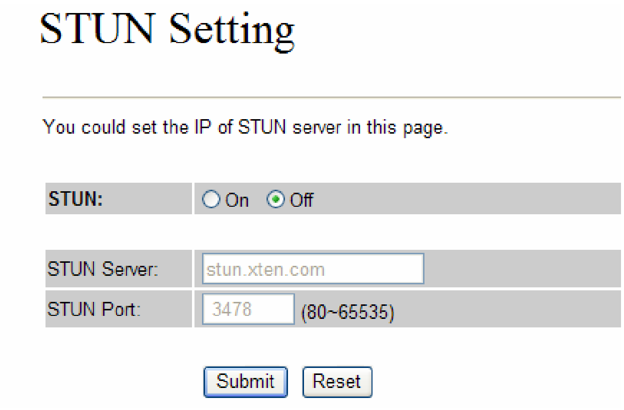
**Keep Alives Period:** To setup the schedule time for delivering the packets.

**Jitter Buffer:** To setup the Jitter Buffer size, and the unit is packet. It needs to refer to the Frame size of Codec.



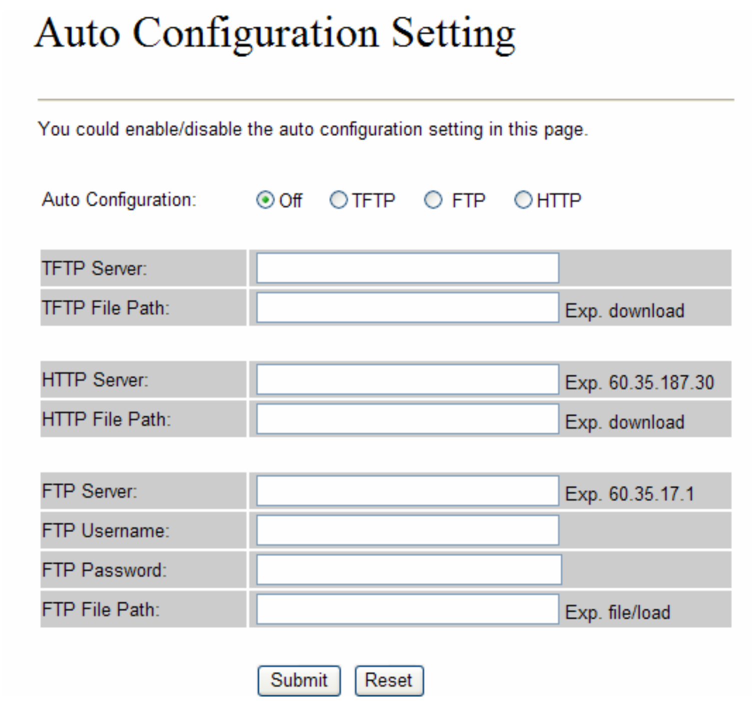

## STUN settings

This page defines the STUN Enable/Disable and STUN Server IP address in this page. This function can help your Phone Adapter working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.



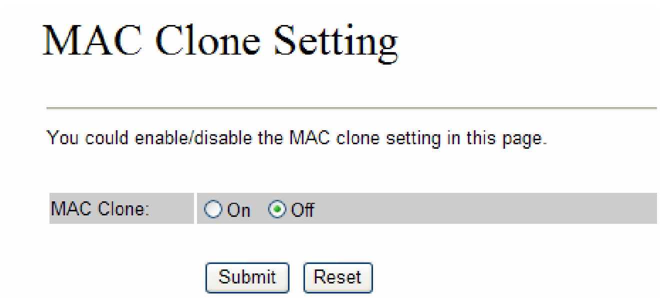

## Auto Configuration

This page defines the Auto Configuration (Auto Provision) setting. IP Phone supports TFTP, FTP, HTTP function in total.



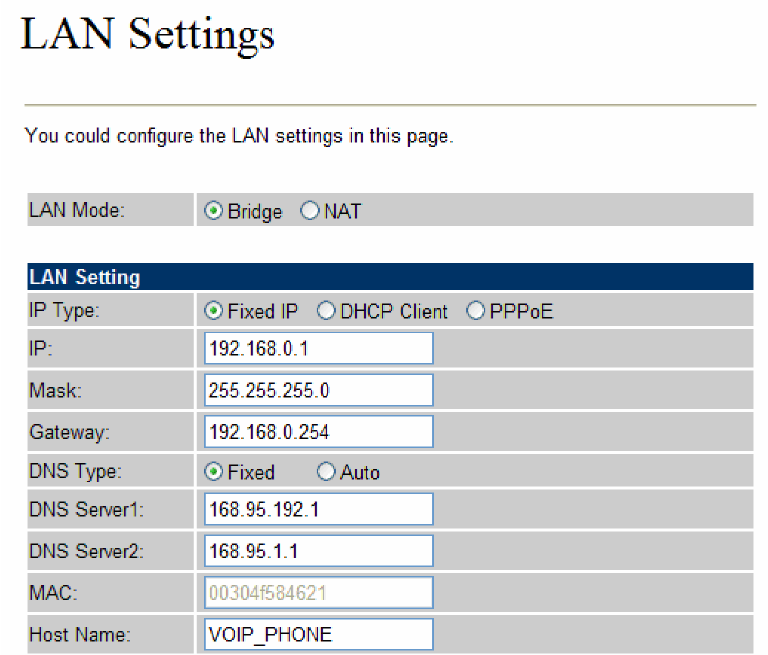

## MAC Clone Setting

This page defines the MAC Clone Enable/Disable. This function will copy the MAC address from NIC (Network Interface Card) which placed in PC to LAN port of IP Phone. That because some ISP will limit the MAC address for PPPoE dial-up connection.

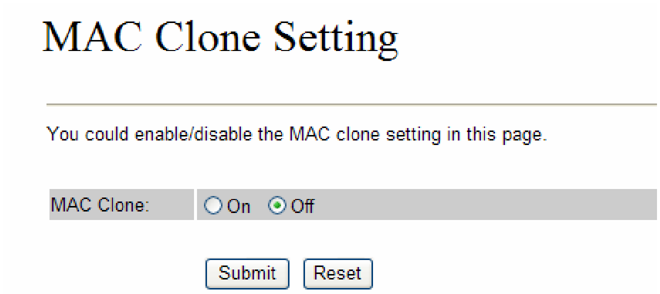


Please refer to the following operate procedures for more understandings to carry out the MAC Clone function.

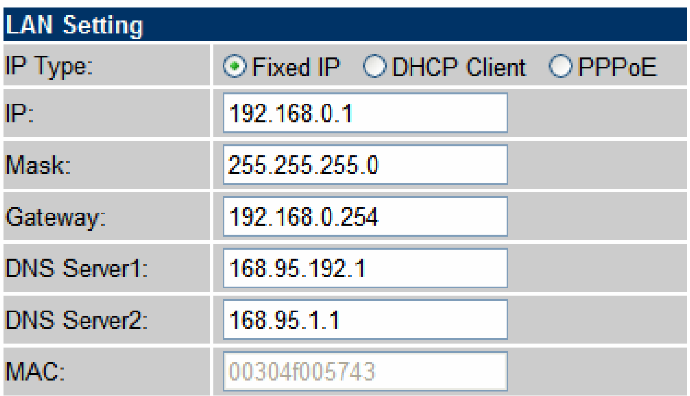
1. Please login IP Phone and browse to “**Network -> LAN Settings**” page. To switch the LAN mode to **NAT** mode then press **Save&Reboot** button to save the settings and reboot machine.




2. Please make sure the network cable of your PC directly connect with PC port of IP Phone, then re-login IP Phone. (In the **NAT** mode, the default IP address of PC port is http://192.168.123.1 )
3. Please browse to “**Advanced Settings -> MAC Clone Setting**” page and enable the MAC Clone function.




4. IP Phone will prompt if sure want to clone the MAC of your PC to the LAN port of IP Phone.
5. After **Save&Reboot**, the MAC of LAN port will become to PC’s original MAC address.



## Tone Settings

This page defines the Tone settings. This function can setup the related parameters of Dial Tone, Ring Back Tone, Busy Tone, Error Tone and Insert Tone. When you finished the setting, please click the Submit button.

### Tones Settings

You could configure your tones settings in this page.

| | Dial Tone | Ring Back Tone | Busy Tone | Congestion Tone | Ring Tone | Call Waitting Tone |
|---|---|---|---|---|---|---|
| Cadence On: | ☐ | ☑ | ☑ | ☑ | ☑ | ☑ |
| Hi-Tone Freq.: | 440 | 480 | 620 | 620 | 480 | 440 |
| Lo-Tone Freq.: | 350 | 440 | 480 | 480 | 440 | 350 |
| Hi-Tone Gain: | 4522 | 2261 | 2261 | 2261 | 15360 | 2261 |
| Lo-Tone Gain: | 2261 | 2261 | 2261 | 2261 | 15360 | 1130 |
| On Time 1: | 0 | 200 | 50 | 30 | 200 | 30 |
| Off Time 1: | 0 | 400 | 50 | 20 | 400 | 20 |
| On Time 2: | 0 | 0 | 0 | 0 | 0 | 30 |
| Off Time 2: | 0 | 0 | 0 | 0 | 0 | 400 |
| On Time 3: | 0 | 0 | 0 | 0 | 0 | 0 |
| Off Time 3: | 0 | 0 | 0 | 0 | 0 | 0 |

Submit Reset

## Advanced Settings

This page defines the advanced functions. When you finished the setting, please click the Submit button.

**ICMP Not Echo:** This function can disable echo when someone ping this device, it can avoid haker try to attack the device.

**Send Anonymous CID:** If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information. (The Registration Server also need support this function)

**Management from WAN:** If enable this function, only WAN be able to connect to the management GUI

**Send Flash event:** There are provide two flash formats: DTMF Event and SIP Info.

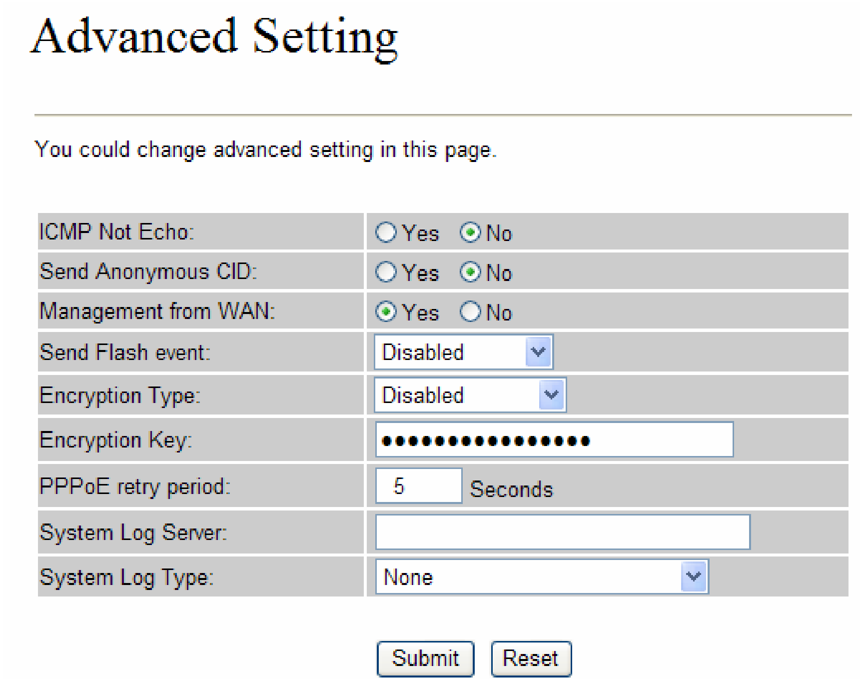
**Encryption Type:** There are provide seven encrypt formats: INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX and VGX. (The Registration Server also need support this function)

**Encryption Key:** The encryption key is use to authentication data transmitted in the SIP network.

**PPPoE retry period:** If PPPoE dial-up connection fail, machine will retry the dial-up motion after this time.

**System Log Server:** Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address.

**System Log Type:** There are seven Syslog types: Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug and All.




## System Authority

In System Authority you can change your login password.


## Save & Reboot

In Save & Reboot you can save the changes you have done. If you want to use new setting in the IP Phone, you have to click the Save button. After you click the Save button, the IP Phone will automatically restart and the new setting will effect.

## Firmware Upgrade

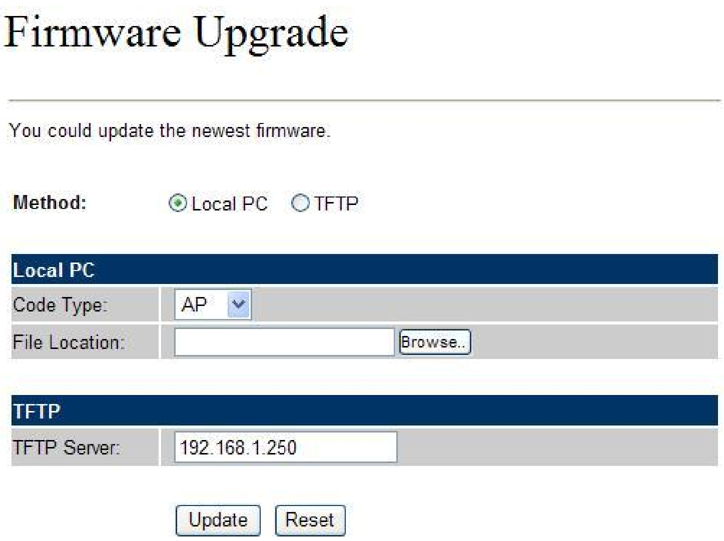
In Firmware Upgrade function you can update new firmware via HTTP or TFTP methods in this page.

You can ugrade the firmware by the following steps:

Select the upgrade method and the firmware code type, AP or DSP code.

Click the “Browse” button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the device then click the Update button.

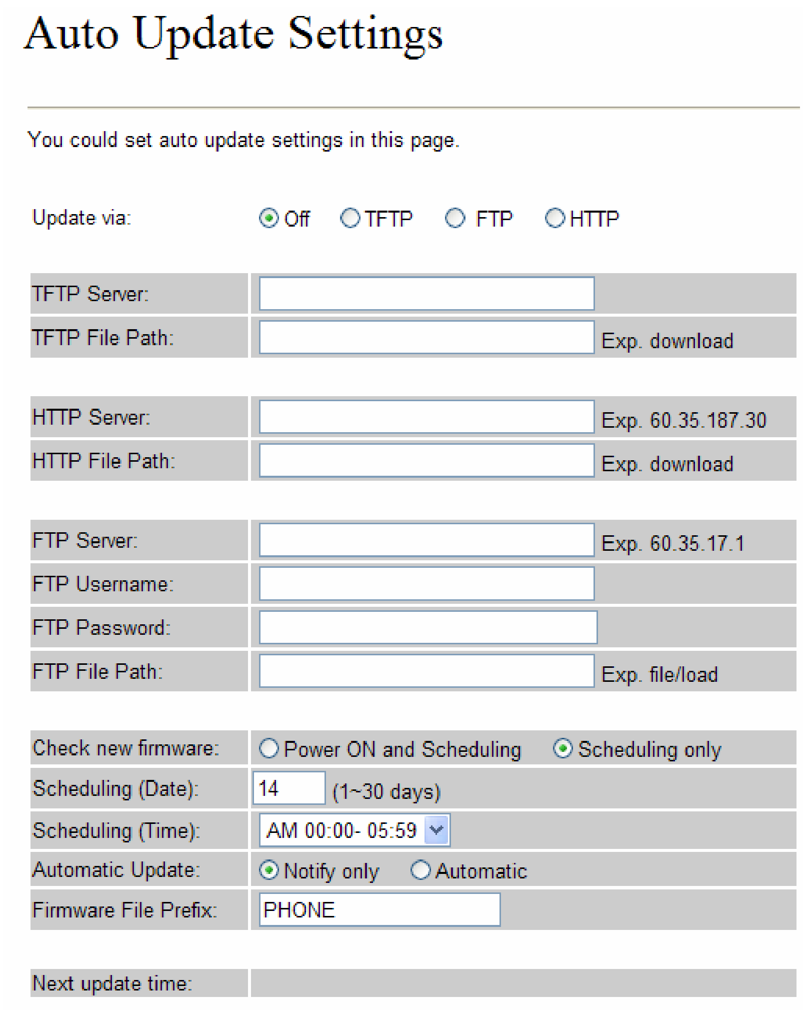



ⓘ**Note**:

After firmware loaded, the unit will be reboot, and Default IP address of the customized firmware: http://192.168.0.1; login name/password: **root/null (no password)**

## Auto Upgrade

The device can update new firmware with the **gz** or **ds** file format automatically by the Auto Upgrade function.




| **Field** | **Descriptions** |
|---|---|
| **Update via** | There are TFTP/ FTP and HTTP three ways to provide the auto upgrade function. |
| **TFTP Server** | Input the TFTP Server address, and it could input the IP or Domain Name form. |
| **TFTP Path** | Set up the file path. |
| **HTTP Server** | Input the HTTP Server address, and it could input the IP or Domain Name form. |
| **HTTP File Path** | Set up the file path. |
| **FTP Server** | Input the FTP Server address, and it could input the IP or Domain Name form. |
| **FTP Username** | The login username. |
| **FTP Password** | The login password |
| **FTP File Path** | Set up the file path. |
| **Check new firmware** | The device will according to the below ways to check the new firmware.<br>**- Power On and Scheduling**: The machine will check the |

| | |
|---|---|
| | new firmware when power on and following the scheduling date and time.<br>**- Scheduling only**: The machine will only follow the scheduling date and time to check the new firmware. |
| **Scheduling (Date)** | The machine will check the new firmware between the time range by random. |
| **Automatic Update** | There are Notify only and Automatic ways to update.<br>**- Notify only**: If there are new firmware, the IP Phone will send the "Be Be Be" sounds when pick up the handset to prompt there are new firmware.<br>**- Automatic**: The device will carry firmware update out automatically. |
| **Firmware File Prefix** | It will check the information of model name. |
| **Next update time** | It will show the next check date and time. |

ⓘ**Note**:

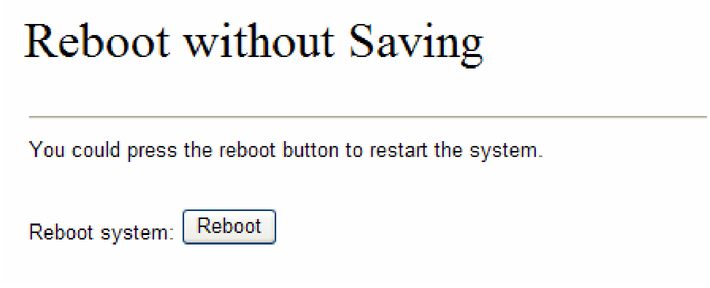
If the **Check new firmware** field selected to Power On, the machine will chck the new firmware accoeding the scheduling time/date and power on. If there are new firmware can be upgraded, the machine won't carry firmware update out automatic. The machine will show the [Found New s/w] message on LCD. Then press [**Menu**] button for entrying the main menu and select the [**7.Administrator -> 2. Upgrade System -> 1.Upgrade Now**] selection to carry out the upgrade firmware action.

## Reset to Default

In Default Setting you can restore the IP Phone to factory default in this page. You can just click the Restore button, then the IP Phone will restore to default and automatically restart again.

## Reboot without saving

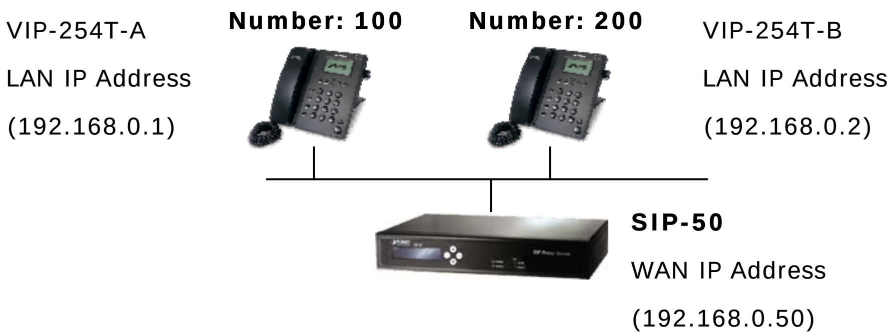
Reboot function you can restart the IP Phone. If you want to restart the IP Phone, you can just click the Reboot button, then the IP Phone will reboot automatically.



# Appendix A Voice communications

There are several ways to make calls to desired destination in IP Phone. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

## Case 1: Voice communication via SIP proxy server _SIP-50



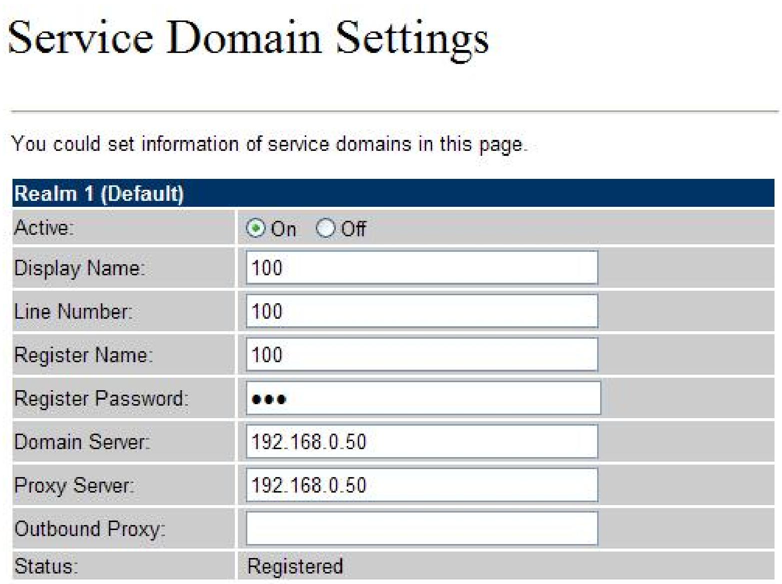

### Machine configuration on the VIP-254T:

**STEP 1:**

Log in SIP-50 and create two testing accounts/password: **100** / **123** (for VIP-254T-A), and **200** / **123** (for VIP-254T-B) for the voice calls.

**STEP 2:**

Please log in VIP-254T-A via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:



**STEP 3:**

Repeat the same configuration steps on VIP-254T-B, and check the machine registration status, make sure the registrations are completed.

**STEP 4:**

To verify the VoIP communication, please pick up the telephone. Dial the destination number to make call between SIP clients. For example, VIP-254T-A (with number 100) with keypad number 200 to VIP-254T-B, or reversely makes calls from SIP client (VIP-254T-B) to the number 100 (VIP-254T-A).

## Case 2: Call Forward Feature_Example 1

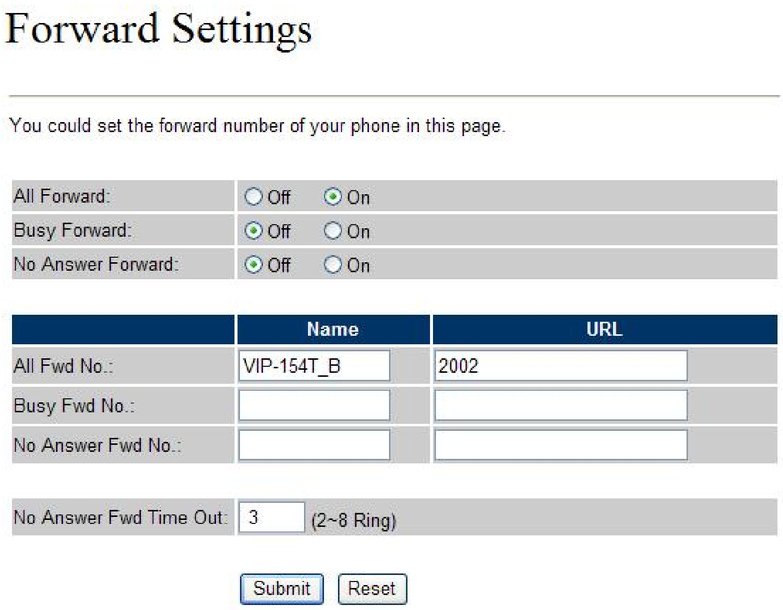
In the following samples, we'll introduce the Call Forward Feature applications.

In this example, there are three VIP-254T register to IPX-300 and VIP-254T_A had set Call Forward function to VIP-254T_B. (The detail registration settings of IPX-300 and VIP-254T please refer to the instruction of Case 3)


### Machine configuration on the VIP-254T:

**STEP 1:**

Please log in VIP-254T_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-254T_B, then the sample configuration screen is shown below:


# Forward Settings

You could set the forward number of your phone in this page.

| | | |
|---|---|---|
| All Forward: | ○ Off | ⦿ On |
| Busy Forward: | ⦿ Off | ○ On |
| No Answer Forward: | ⦿ Off | ○ On |

| | Name | URL |
|---|---|---|
| All Fwd No.: | VIP-154T_B | 2002 |
| Busy Fwd No.: | | |
| No Answer Fwd No.: | | |

| | |
|---|---|
| No Answer Fwd Time Out: | 3 (2~8 Ring) |

Submit Reset

**STEP 2:**

After set up completed and reboot machine, the LCD screen will show below:

| |
|---|
| 10-19 17:20<br># Forward # |

After 2~3 seconds, the LCD screen will show below:

| |
|---|
| 10-19 17:20<br>AF 2002 |

**Test the scenario:**

VIP-254T_C pick up the telephone and dial the number 1001(VIP-254T_A), because VIP-254T_A had set up **All Forward** function to the number 2002(VIP-254T_B), so the number 2002(VIP-254T_B) will ring up then it pick up the telephone and communication with the number 3003(VIP-254T_C).

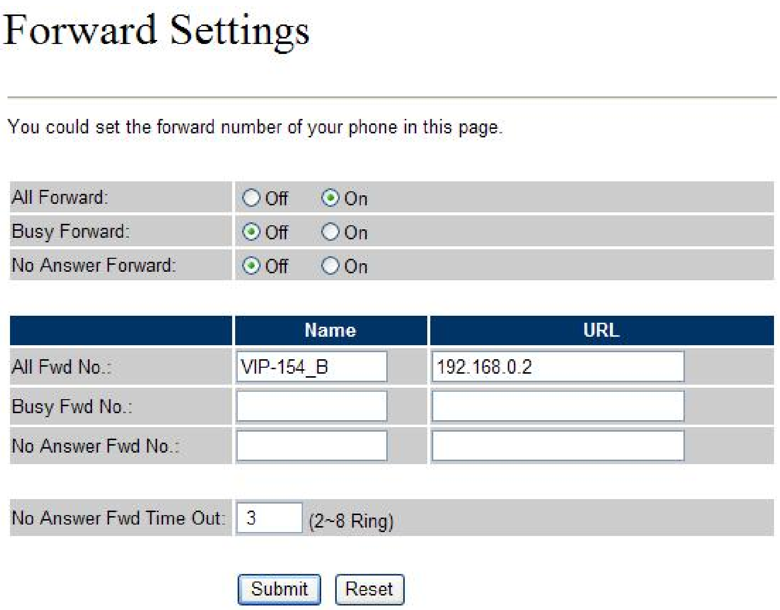
## Case 3: Call Forward Feature_Example 4

In this example, there are three VIP-254T and connect with Peer to Peer mode. VIP-254T_A had set Call Forward function to VIP-254T_B.

## Machine configuration on the VIP-254T:

**STEP 1:**

Please log in VIP-254T_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-254T_B, and then the sample configuration screen is shown below:




**STEP 2:**

After set up completed and reboot machine, the LCD screen will show below:

| |
|---|
| 10-19 17:20 |
| # Forward # |

After 2~3 seconds, the LCD screen will show below:

| |
|---|
| 10-19 17:20 |
| AF 192.168.0.2 |

## Test the scenario:

VIP-254T_C pick up the telephone and dial the IP Address 192.168.0.1(VIP-254T_A), because VIP-254T_A had set up **All Forward** function to the IP Address 192.168.0.2(VIP-254T_B), so the IP Address 192.168.0.2 (VIP-254T_B) will ring up then it pick up the telephone and communication with the VIP-254T_C.

# Appendix B The method of operation guide

In this section, we'll introduce the features method of operation, and lead you step by step to establish these features.

## Call Transfer

### A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A press "**FLASH**" button to hold the conversation with B, and input the number of C (Follow by the "**#**" key).
3. C will ring up, and A hang up the handset.
4. C picks up the handset and conversation with B.

### B. Attendant Transfer

1. B call to A and they are in the process of conversation.
2. A press "**FLASH**" button to hold the conversation with B, and input the number of C (Follow by the "**#**" key).
3. C will ring up.
4. C picks up the handset and conversation with A.
5. A hang up and C conversation with B.

## 3-Way Conference

1. A and B are in the process of conversation.
2. A want to invite C to join their conversation.
3. A press "**FLASH**" button to hold the conversation with B, and input the number of C (Follow by the "**#**" key).
4. C will ring up and pick up the handset to conversation with A.
5. A press "**CONF**" button and they will entry the 3-Way conference mode.

## Call Waiting

1. A and B are in the process of conversation.
2. C call to A and A will hear the prompt sounds.
3. A press "**Hold**" button to hold the conversation with B, and switch to conversation with C.

## Switch the Realm (Registration Proxy Server)

IP Phone can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input the switch code.

**Realm switch code:**

**1***: Realm 1

**2***: Realm 2

**3***: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

# Appendix C VIP-254T / VIP-254PT Specifications

| | | |
|---|---|---|
| Product | SIP IP Phone | SIP PoE IP Phone |
| Model | VIP-254T | VIP-254PT |
| Hardware | | |
| LAN | 1 x 10/100Mbps RJ-45 port<br>Power Over Ethernet 802.3af compliant at VIP-254PT | |
| PC | 1 x 10/100Mbps RJ-45 port | |
| LCD display | 2 x 16 characters | |
| Speaker | Full duplex hands free speaker phone | |
| Protocols and Standard | | |
| Standard | SIP 2.0 (RFC3261), MD5 for SIP authentication (RFC2069/ RFC 2617), SIP outbound proxy, SIP NAT Traversal Support STUN (RFC3489) | |
| Voice codec | G.711: 64k bit/s (PCM)<br>G.723.1: 6.3k / 5.3k bit/s<br>G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)<br>G.729A: 8k bit/s (CS-ACELP)<br>G.729B: adds VAD & CNG to G.729 | |
| Voice Standard | Voice activity detection (VAD)<br>Comfort noise generation (CNG)<br>Acoustic echo canceller (AEC)<br>G.165: Line echo canceller (LEC)<br>Jitter Buffer | |
| Supplementary services | Caller ID<br>3-way conference<br>Immediate (unconditional) call forwarding<br>Busy call forwarding<br>No answer calls forwarding<br>Call Hold/Waiting/Transferring | |
| Call history | Record incoming call<br>Outgoing call<br>Missed (not accepted) call history | |
| Protocols | SIP v1 (RFC2543), v2(RFC3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, RARP, DNS, DHCP, SNTP, PPPoE | |
| Network and Configuration | | |
| Access Mode | Static IP, PPPoE, DHCP | |
| Management | Web, LCD menu keypad, auto-provision by TFTP/FTP/HTTP | |
| Dimension (W x D x H) | 184 mm x 200 mm x 48 mm | |
| Operating Environment | 0~50 degree C, 0~90% humidity | |
| Power Requirement | 5V DC, 1A<br>Power Over Ethernet 802.3af compliant at VIP-254PT | |
| EMC/EMI | CE, FCC Class B | |

# EC Declaration of Conformity

For the following equipment:

*Type of Product : SIP IP Phone
*Model Number : VIP-254T

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is hereby confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive (2004/108/EC), For the evaluation regarding the Electromagnetic Compatibility (2004/108/EC), the following standards are applied:

Emission
EN55022: 1998 + A1: 2000 + A2: 2003
EN61000-3-2: 2000 + A2: 2005
EN61000-3-3: 1995 + A1: 2001 + A2: 2005
EN55024: 1998 + A1: 2001 + A2: 2003
- IEC 61000-4-2: 1995 + A1: 1998 + A2: 2000
- IEC 61000-4-3: 2002 + A1: 2002
- IEC 61000-4-4: 2004
- IEC 61000-4-5: 1995 + A1: 2000
- IEC 61000-4-6: 1996 + A1: 2000
- IEC 61000-4-8: 1993 + A1: 2000
- IEC 61000-4-11: 2004

**Responsible for marking this declaration if the:**

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

**Authorized representative established within the EU (if applicable):**

**Company Name:** **Planet Technology Corp.**

**Company Address:** **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

**Person responsible for making this declaration**

**Name, Surname** **Jonas Yang**

**Position / Title :** **Product Manager**

Jonas

| | | |
|---|---|---|
| **Taiwan** | **12 May, 2007** | |
| *Place* | *Date* | *Legal Signature* |

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PLANET
Networking & Communication

# EC Declaration of Conformity

For the following equipment:

*Type of Product : SIP PoE IP Phone
*Model Number : VIP-254PT

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is hereby confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive (2004/108/EC), For the evaluation regarding the Electromagnetic Compatibility (2004/108/EC), the following standards are applied:

Emission
EN55022: 1998 + A1: 2000 + A2: 2003
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EN61000-3-3: 1995 + A1: 2001 + A2: 2005
EN55024: 1998 + A1: 2001 + A2: 2003
IEC 61000-4-2: 1995 + A1: 1998 + A2: 2000
IEC 61000-4-3: 2002 + A1: 2002
IEC 61000-4-4: 2004
IEC 61000-4-5: 1995 + A1: 2000
IEC 61000-4-6: 1996 + A1: 2000
IEC 61000-4-8: 1993 + A1: 2000
IEC 61000-4-11: 2004

**Responsible for marking this declaration if the:**

**☒ Manufacturer ☐ Authorized representative established within the EU**

**Authorized representative established within the EU (if applicable):**

**Company Name: Planet Technology Corp.**

**Company Address: 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

**Person responsible for making this declaration**

**Name, Surname Jonas Yang**

**Position / Title : Product Manager**

| Taiwan | 12 May, 2008 | Jonas |
|---|---|---|
| *Place* | *Date* | *Legal Signature* |


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